

PCT

WORLD INTELLECTUAL PROPERTY ORGANIZATION  
International Bureau



INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : <b>G10L 3/00</b>		<b>A1</b>	(11) International Publication Number: <b>WO 96/32710</b>
			(43) International Publication Date: 17 October 1996 (17.10.96)
(21) International Application Number: PCT/US96/04974		(81) Designated States: AL, AM, AT, AU, AZ, BB, BG, BR, BY, CA, CH, CN, CZ, DE, DK, EE, ES, FI, GB, GE, HU, IS, JP, KE, KG, KP, KR, KZ, LK, LR, LS, LT, LU, LV, MD, MG, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, TJ, TM, TT, UA, UG, US, UZ, VN, ARIPO patent (KE, LS, MW, SD, SZ, UG), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, ML, MR, NE, SN, TD, TG).	
(22) International Filing Date: 10 April 1996 (10.04.96)			
(30) Priority Data: 08/419,200 10 April 1995 (10.04.95) US 08/420,721 10 April 1995 (10.04.95) US			
(71) Applicant (for all designated States except US): CORPORATE COMPUTER SYSTEMS, INC. [US/US]; Building #4, 670 North Beers Road, Holmdel, NJ 07733 (US).			
(72) Inventor; and (75) Inventor/Applicant (for US only): HINDERKS, Larry, W. [US/US]; 37 Ladwood Drive, Holmdel, NJ 07733 (US).			
(74) Agents: SMALL, Dean, D. et al.; McAndrews, Held & Malloy, Ltd., Suite 3400, 500 West Madison, Chicago, IL 60661 (US).			

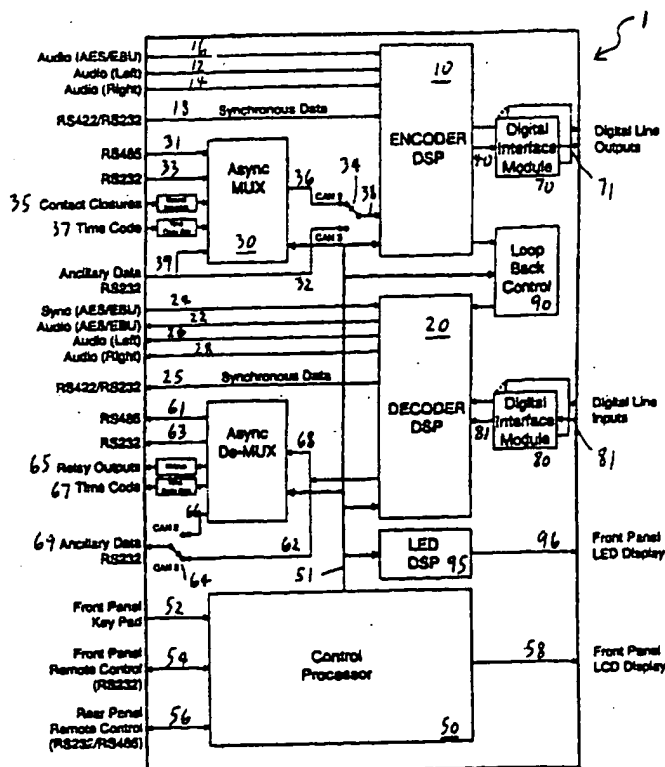
Published

With international search report.  
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: SYSTEM FOR COMPRESSION AND DECOMPRESSION OF AUDIO SIGNALS FOR DIGITAL TRANSMISSION

(57) Abstract

Ancillary data (39) may be multiplexed (30) and encoded (10) with audio data (12, 14) and transmitted (70, 71) in such a way that it may be decoded (20) when received (80, 81).



**FOR THE PURPOSES OF INFORMATION ONLY**

Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

AM	Armenia	GB	United Kingdom	MW	Malawi
AT	Austria	GE	Georgia	MX	Mexico
AU	Australia	GN	Guinea	NE	Niger
BB	Barbados	GR	Greece	NL	Netherlands
BE	Belgium	HU	Hungary	NO	Norway
BF	Burkina Faso	IE	Ireland	NZ	New Zealand
BG	Bulgaria	IT	Italy	PL	Poland
BJ	Benin	JP	Japan	PT	Portugal
BR	Brazil	KE	Kenya	RO	Romania
BY	Belarus	KG	Kyrgyzstan	RU	Russian Federation
CA	Canada	KP	Democratic People's Republic of Korea	SD	Sudan
CF	Central African Republic	KR	Republic of Korea	SE	Sweden
CG	Congo	KZ	Kazakhstan	SG	Singapore
CH	Switzerland	LI	Liechtenstein	SI	Slovenia
CI	Côte d'Ivoire	LK	Sri Lanka	SK	Slovakia
CM	Cameroon	LR	Liberia	SN	Senegal
CN	China	LT	Lithuania	SZ	Swaziland
CS	Czechoslovakia	LU	Luxembourg	TD	Chad
CZ	Czech Republic	LV	Latvia	TG	Togo
DE	Germany	MC	Monaco	TJ	Tajikistan
DK	Denmark	MD	Republic of Moldova	TT	Trinidad and Tobago
EE	Estonia	MG	Madagascar	UA	Ukraine
ES	Spain	ML	Mali	UG	Uganda
FI	Finland	MN	Mongolia	US	United States of America
FR	France	MR	Mauritania	UZ	Uzbekistan
GA	Gabon			VN	Viet Nam

SYSTEM FOR COMPRESSION AND DECOMPRESSION  
OF AUDIO SIGNALS FOR DIGITAL TRANSMISSION

RELATED APPLICATION

The present application relates to co-pending PCT application  
5 \_\_\_\_\_, filed April 10, 1996, entitled "Method and  
Apparatus for Transmitting Coded Audio Signals Through a  
Transmission Channel With Limited Bandwidth" by the same inventor  
and assigned to the Assignee of the present application. The co-  
pending PCT application noted above is incorporated by reference in  
10 its entirety along with any appendices and attachments thereto.

SOURCE CODE APPENDIX

Source code for the control processor of the present invention  
has been included as a SOURCE CODE APPENDIX.

FIELD OF THE INVENTION

15 The present invention relates generally to an audio CODEC for  
the compression and decompression of audio signals for transmission  
over digital facilities, and more specifically, relates to an audio  
CODEC that is programmable by a user to control various CODEC  
operations, such as monitoring and adjusting a set of psycho-  
20 acoustic parameters, selecting different modes of digital  
transmission, and downloading new compression algorithms.

BACKGROUND OF THE INVENTION

Current technology permits the translation of analog audio signals into a sequence of binary numbers (digital). These numbers may then be transmitted and received through a variety of means.

5 The received signals may then be converted back into analog audio signals. The device for performing both the conversion from analog to digital and the conversion from digital to analog is called a CODEC. This is an acronym for Coder/DECoder.

10 The cost of transmitting bits from one location to another is a function of the number of bits transmitted per second. The higher the bit transfer rate the higher the cost. Certain laws of physics in human and audio perception establish a direct relationship between perceived audio quality and the number of bits transferred per second. The net result is that improved audio  
15 quality increases the cost of transmission.

CODEC manufacturers have developed technologies to reduce the number of bits required to transmit any given audio signal (compression techniques) thereby reducing the associated transmission costs. The cost of transmitting bits is also a  
20 function of the transmission facility used, i.e., satellite, PCM phone lines, ISDN (fiber optics).

A CODEC that contains some of these compression techniques also acts as a computing device. It inputs the analog audio signal, converts the audio signal to a digital bit stream, and then  
25 applies a compression technique to the bit stream thereby reducing



the number of bits required to successfully transmit the original audio signal. The receiving CODEC applies the same compression techniques in reverse (decompression) so that it is able to convert the compressed digital bit stream back into an analog audio signal.

5 The difference in quality between the analog audio input and the reconstructed audio output is an indication of the quality of the compression technique. The highest quality technique would yield an identical signal reconstruction.

10 Currently, the most successful compression techniques are called perceptual coding techniques. These types of compression techniques attempt to model the human ear. These compression techniques are based on the recognition that much of what is given to the human ear is discarded because of the characteristics of the ear. For example, if a loud sound is presented to a human ear  
15 along with a softer sound, the ear will only hear the loud sound. As a result, encoding compression techniques can effectively ignore the softer sound and not assign any bits to its transmission and reproduction under the assumption that a human listener can not hear the softer sound even if it is faithfully transmitted and  
20 reproduced.

Many conventional CODECs use perceptual coding techniques which utilize a basic set of parameters which determine their behavior. For example, the coding technique must determine how soft a sound must be relative to a louder sound in order to make  
25 the softer sound a candidate for exclusion from transmission. A

number which determines this threshold is considered a parameter of the scheme which is based on that threshold. These parameters are largely based on the human psychology of perception so they are collectively known as psycho-acoustic parameters.

5        However, conventional CODECs which use perceptual coding have experienced limitations. More specifically, manufacturers of existing CODECs preprogram all of the CODEC's operating variables which control the compression technique, decompression technique, bit allocation and transmission rate. By preprogramming the CODEC,  
10       the manufacturer undesirable limits the user interaction with the resulting CODEC. For example, it is known that audio can be transmitted by digital transmission facilities. These digital transmissions include digital data services, such as conventional phone lines, ISDN, T1, and E1. Other digital transmission paths  
15       include RF transmission facilities such as spread spectrum RF transmission and satellite links.

         Although existing CODECs can transmit compressed audio signals via digital transmission facilities, any variables regarding the mode of transmission are preprogrammed by the manufacturer of the  
20       CODEC, thereby limiting the CODEC's use to a single specific transmission facility. Hence, the user must select a CODEC which is preprogrammed to be compatible with the user's transmission facility. Moreover, existing CODECs operate based on inflexible compression and bit allocation techniques and thus, do not provide  
25       users with a method or apparatus to monitor or adjust the CODEC to

fit the particular user's wants and needs. Accordingly, users must test CODECs with different compression and bit allocation techniques and then select the one device which has the features or options so desired, e.g. satellite transmission capabilities.

5       Moreover, standard coding techniques have been developed in order to ensure interoperability of CODECs from different manufacturers and to ensure an overall level of audio quality, thereby limiting the CODEC's use to a single specific transmission facility. One such standard is the so-called ISO/MPEG Layer-II  
10       compression standard, for the compression and decompression of an audio input. This standard sets forth a compression technique and a bit stream syntax for the transmission of compressed binary data. The ISO/MPEG Layer-II standard defines a set of psycho-acoustic parameters that is useful in performing compression. U.S. Patent  
15       No. 4,972,484, entitled "Method of Transmitting or Storing Masked Sub-band Coded Audio Signals," discloses the ISO/MPEG Layer-II standard and is incorporated by reference.

          However, conventional CODECs do not use a uniform set of parameters. Each CODEC manufacturer determines their own set of  
20       psycho-acoustic parameters either from a known standard or as modified by the manufacturer in an attempt to provide the highest quality sound while using the lowest number of bits to encode audio. Once the manufacturer selects a desired parameter set, the manufacturer programs values for each of the parameters. These

preprogrammed parameter values correspond to the manufacturer's perception of an optimal audio quality at the decoder.

However, in conventional CODECs, users typically are unaware of the existence or nature of these parameters. Further, the user has no control over the parameter values. As a result, users were required to test different CODECs from different manufacturers and then select the CODEC that met the user's requirements or that sounded best to the user.

Typically, conventional CODECs utilize standard parameters which have been accepted by the International Standards Organization (ISO) and have been adopted as part of the International Standards Organization, Motion Picture Experts Group (ISO/MPEG) Layer-II compression standard. However, the ISO/MPEG Layer-II standard has met with limited acceptance since these parameters do not necessarily provide CD quality output. The ISO/MPEG Layer-II parameters are determined and set based on the average human ear. The parameters do not account for the variations between each individual's hearing capabilities. Hence, the conventional standards and CODECs do not afford the ability for users to tune their CODEC to the user's individual subjective hearing criteria. Nor are conventional CODECs able to meet changing audio needs and to shape the overall sound of their application.

A need remains within the industry for an improved CODEC which is more flexible, programmable by the user, and which overcomes the

disadvantages experienced heretofore. It is an object of the present invention to meet this need.

#### OBJECTS OF THE INVENTION

5 It is an object of the present invention to provide a programmable audio CODEC that can be monitored, controlled and adjusted by a user to control the various functions of the CODEC.

10 It is another object of the present invention to provide an audio CODEC that is programmable by a user to transmit compressed digital bit streams over various user selected digital transmission facilities.

It is an object of the present invention to provide a user programmable audio CODEC with a plurality of psycho-acoustic parameters that can be monitored, controlled, and adjusted by a user to change the audio output from the CODEC.

15 It is a related object of the present invention to provide an audio CODEC with new psycho-acoustic parameters.

It is a further related object of the present invention to provide an audio CODEC where the psycho-acoustic parameters are changed by knobs on the front panel of the CODEC.

20 It is another related object of the present invention to provide an audio CODEC where the psycho-acoustic parameters are changed by a keypad on the front panel of the CODEC.

It is still a further related object of the present invention to provide an audio CODEC with a personal computer connected

thereto to adjust the psycho-acoustic parameters by changing graphic representations of the parameters on a computer screen.

5 It is a related object of the present invention to provide an audio CODEC that is programmable by a user to transmit compressed digital bit streams over a digital data service.

It is a further related object of the present invention to provide an audio CODEC that is programmable by a user for transmission of compressed digital bit streams over any of T1, E1 and ISDN lines or over RF transmission facilities.

10 It is yet another related object of the present invention to provide an audio CODEC that is user programmable for transmission of compressed digital bit streams via satellite.

15 It is a further object of the present invention to provide an audio CODEC for transmission of asynchronous data together with the transmission of compressed audio.

It is still a further object of the present invention to provide an audio CODEC that utilizes the multiple audio compression and decompression schemes.

20 It is still another object of the present invention to provide an audio CODEC which allows a user to select one of several stored audio compression techniques.

It is still another object of the present invention to provide an audio CODEC that is remotely controlled by a host computer.

It is still another object of the present invention to provide an audio CODEC for monitoring either the encoder input signal or the decoder output signal with the use of headphones.

5 It is still another object of the present invention to provide an audio CODEC with safeguards for automatically selecting a second transmission facility if a first user selected transmission facility fails.

10 It is yet another object of the present invention to provide an audio CODEC that can be controlled by inputting control commands into a key pad on the front panel of the CODEC.

It is related object of the present invention to provide an audio CODEC having a user interface to control and program the audio CODEC through the use of a graphics display on the front panel.

15 It is still another related object of the present invention to provide for connection of a personal computer to the audio CODEC for controlling the input of program information thereto.

It is still another object of the present invention to provide bi-directional communication between two audio CODECs.

20 It is still another object of the present invention to provide an audio CODEC that can be interfaced to a local area network.

It is yet another object of the present invention to provide an audio CODEC that will provide programmed information to users through the use of indicators on the front panel of the CODEC.

It is yet another object of the present invention to provide an audio CODEC that can send non-audio compressed information including text, video and graphic information.

5 It is still another object of the present invention to provide an audio CODEC that can store and retrieve information on and from an electronic storage medium or a disk drive.

It is still another related object of the present invention to provide an audio CODEC that can transmit control information along with the textual video and graphic information.

10 It is still a further object of the present invention to provide digital audio compression techniques that yield improved and preferably CD quality audio.

It is a related object of the present invention to provide a compression scheme that yields better audio quality than the MPEG  
15 compression standard.

It is still another related object of the present invention to provide CD quality audio that achieves a 12 to 1 compression ratio.

#### SUMMARY OF THE INVENTION

20 The present invention provides a CODEC which holds several compression algorithms and allows the user easily to download future audio compression algorithms as needed. This makes the present CODEC very versatile and prevents it from becoming obsolete.



The preferred CODEC provides for both digital and analog input of external signals. The CODEC is also capable of handling a wide variety of ancillary data which can be incorporated into the compressed bit stream along with the audio and header data. The ancillary bit stream preferably enters the encoder directly from external sources. However, the user could alternatively choose to have the external data multiplexed into a composite ancillary bit stream before being encoded with the audio and header data. The preferred CODEC also provides for rate adaptation of signals that are input (and output) at one rate and compressed (and decompressed) at yet another rate. This rate adaptation can also be synchronized to external clock sources.

The user can also programmably alter the psycho-acoustic compression parameters to optimize transmissions under different conditions. The disclosed invention also allows the user to programmably control CODEC transmission modes as well as other CODEC operations. Such programmable control is achieved through remote interfaces and/or direct keypad control.

The compressed output signal can also be interfaced with a variety of external sources through different types of output Digital Interface Modules (DIMs). Similar input DIMs would input return signals for decoding and decompression by the CODEC. Certain specialized DIMs might also operate as satellite receiver modules. Such modules would preferably store digital information as it becomes available for later editing and use. Satellite

receiver modules would be capable of receiving information such as audio, video, text, and graphics. This information would then be decoded and decompressed as appropriate by the CODEC.

5 Additional features and advantages of the present invention will become apparent to one of skilled in the art upon consideration of the following detailed description of the present invention.

#### BRIEF DESCRIPTIONS OF THE DRAWINGS

10 Figure 1 is a block diagram of a CODEC illustrating signal connections between various components in accordance with a preferred embodiment of the present invention.

Figure 2 is a block diagram of a CODEC illustrating signal connections between various components in accordance with the preferred embodiment shown in Figure 1.

15 Figure 3 is a block diagram illustrating ancillary data being multiplexed into a composite bit stream in accordance with the preferred embodiment of Figure 1.

20 Figure 4 is a block diagram illustrating an ISO/MPEG audio bit stream being decoded into a composite ancillary bit stream and audio left and right signals in accordance with the preferred embodiment of Figure 1.

Figure 5 is an example of a front panel user keypad layout in accordance with a preferred embodiment of the present invention.

Figure 6 is another example of a front panel user keypad layout in accordance with a preferred embodiment of the present invention.

5 Figure 7 is another example of a front panel user keypad layout in accordance with a preferred embodiment of the present invention.

10 Figure 8 is a block diagram showing the decoder output timing with the AES/EBU sync disabled or not present and using normal timing in accordance with a preferred embodiment of the present invention.

Figure 9 is a block diagram showing the decoder output timing with AES/EBU sync disabled or not present using internal crystal timing in accordance with a preferred embodiment of the present invention.

15 Figure 10 is a block diagram showing decoder output timing with AES/EBU sync enabled and present using AES timing in accordance with a preferred embodiment of the present invention.

Figure 11 is an example of an LED front panel display in accordance with a preferred embodiment of the present invention.

20 Figure 12 is another example of an LED front panel display in accordance with a preferred embodiment of the present invention.

Figure 13 is a block diagram of a CODEC illustrating signal connections between various components allowing transmission of audio, video, text, and graphical information in accordance with a preferred embodiment of the present invention.

25

Figure 14 is a diagram illustrating the interconnection between various modules in accordance with a preferred embodiment.

Figure 15 is a block diagram of an embodiment of an encoder as implemented in the CODEC of the system in accordance with the preferred embodiment shown in Figure 14.

Figure 16 is a diagram illustrating a known representation of a tonal masker as received and recognized by a CODEC system.

Figure 17 is a diagram illustrating a known representation of a tonal masker and its associated masking skirts as recognized by a CODEC system.

Figure 18 is a diagram illustrating a tonal masker and its associated masking skirts as implemented by the encoder of the system in accordance with the preferred embodiment shown in Figure 14.

Figure 19 is a diagram illustrating the representation of the addition of two tonal maskers as implemented by the encoder of the system in accordance with the preferred embodiment shown in Figure 14.

Figure 20 is a block diagram illustrating the adjustment of a single parameter as performed by the encoder of the system in accordance with the preferred embodiment shown in Figure 14.

Figure 21 illustrates a block diagram of an encoder for a single audio channel according to the present invention.

Figure 22 illustrates a data structure used in the preferred embodiment for a frame of data.

Figure 23 illustrates a block diagram of an encoder for two audio channels operated in joint stereo according to the present invention.

5 Figure 24 illustrates a flow diagram of the process followed by the present invention when adjusting the scaling factors.

Figures 25a and 25b illustrate a flow diagram of the overall process followed by the present invention when assigning encoding levels to the quantizers.

10 Figure 26 illustrates a flow diagram of the process followed by the present invention when obtaining a global masking threshold.

Figure 27 illustrates a flow diagram of the process followed by the present invention predicting bit allocation for mono, stereo or joint stereo frames.

15 Figure 28 illustrates a flow diagram of the process followed by the present invention when determining an allocation step for a specific subband.

Figure 29 illustrates a flow diagram of the process followed by the present invention when determining the joint stereo boundary.

20 Figure 30 illustrates a flow diagram of the process followed by the present invention when assigning a quantization level.

Figure 31 illustrates a flow diagram of the process followed by the present invention when deallocating bits from one or more subbands following the initial allocation process.

Figures 32a and 32b illustrate graphs of exemplary subbands having a portion of the global masking threshold therein and multiple masking-to-noise ratios therein corresponding to multiple allocation steps.

5        Figure 33 illustrates a deallocation table recorded during bit allocation and de-allocation.

#### DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

##### **CODEC System with Adjustable Parameters**

10        With reference to FIGURES 14 and 15, a CODEC 1010 has an encoder 1012 and a decoder 1010. The encoder 1012 receives as input an analog audio source 1016. The analog audio source 1016 is converted by an analog to digital converter 1018 to a digital audio bit stream 1020. The analog to digital converter 1018 can be located before the encoder 1012, but is preferably contained  
15        therein. In the encoder 1012, compression techniques compress the digital audio bit stream 1020 to filter out unnecessary and redundant noises. In the preferred embodiment, the compression technique utilizes the parameters defined by the ISO/MPEG Layer-II standard as described in USP 4,972,484, and in a document entitled,  
20        "Information Technology Generic Coding Of Moving Pictures And Associated Audio," and is identified by citation ISO 3-11172 Rev. 2. The '484 patent and the ISO 3-11172, Rev. 2 Document are incorporated by reference.

In addition, the compression technique of the preferred embodiment of the encoder 1012 adds several new parameters as explained below. The resultant compressed digital audio bit stream 1022 is then transmitted by various transmission facilities (not shown) to a decoder at another CODEC (not shown). The decoder decompresses the digital audio bit stream and then the digital bit stream is converted to an analog signal.

The compression technique utilized by the CODEC 1010 to compress the digital audio bit stream 1020 is attached as the Source Code Appendix, and is hereby incorporated by reference.

#### **Human Auditory Perception - Generally**

The audio compression routine performed by the encoder 1012 is premised on several phenomena of human auditory perception. While those phenomena are generally understood and explained in the ISO Document and '484 patent referenced above, a brief summary is provided hereafter.

Generally, it is understood that when a human ear receives a loud sound and a soft sound, close in time, the human will only perceive the loud sound. In such a case, the loud sound is viewed as "masking" or covering up the quiet or soft sound.

The degree to which the softer sound is masked is dependent, in part, upon the frequencies of the loud and soft sounds and the distance between the frequencies of the loud and soft sounds. For instance, a loud sound at 700 Hz will have a greater masking effect

upon a soft sound at 750 Hz than upon a soft sound at 900 Hz. Further, typically, the ear is more discriminating between loud and soft sounds at low frequencies as compared to loud and soft sounds at high frequencies.

5        Another aspect of hearing and psycho-acoustics is that a person can hear two tones at the same frequency provided that the softer tone is close enough in amplitude to the louder tone. The maximum difference in amplitude between the two tones of common frequency is referred to as the masking index. The masking index  
10       is dependent, in part, upon frequency of the tones. Generally, the masking index increases with frequency. For instance, the masking index of a masking tone at 1000 Hz will be smaller than the masking index of a masking tone at 7000 Hz.

FIGURE 17 illustrates the masking index 1034 for the tonal  
15       masker 1024. Thus, the masking effect will be greater for a loud sound at 7000 Hz upon a soft sound 7050 Hz as compared to the masking effect of a loud sound at 700 Hz upon a soft sound at 750 Hz. The masking effect of a sound is defined by its "masking skirt," which is explained below.

20       The encoder defines maskers and masking skirts based on the above noted masking effects (as explained below in more detail). If masking does occur, then the compression technique will filter out the masked (redundant) sound.

      The audio compression technique of the encoder is also  
25       premised on the assumption that there are two kinds of sound



maskers. These two types of sound maskers are known as tonal and noise maskers. A tonal masker will arise from audio signals that generate nearly pure, harmonically rich tones or signals. A tonal masker that is pure (extremely clear) will have a narrow bandwidth.

5 The band width of a tonal masker varies with frequency. In particular, tones at high frequency may have a wider bandwidth than low frequency tones. For instance, a sound centered at 200 Hz with a width of 50 Hz may not be considered a tone, while a sound centered at 7000 Hz with a width of 200 Hz could be considered a

10 tone. Many sounds have no single dominant frequency (tonal), but instead are more "noise" like. If a sound is wide in bandwidth, with respect to its center frequency, then the sound is classified as noise and may give rise to a noise masker. A noise masker will arise from signals that are not pure. Because noise maskers are

15 not pure, they have a wider bandwidth and appear in many frequencies and will mask more than the tonal masker.

FIGURE 16 illustrates a tonal masker 1024 as a single vertical line at a frequency which remains constant as the power increases to the peak power 1026. By way of example, the tonal masker may

20 have 46 HZ bandwidth. Sounds within that bandwidth, but below the peak power level 1026 are "masked." An instrument that produces many harmonics, such as a violin or a trumpet, may have many such tonal maskers. The method for identifying tonal maskers and noise maskers is described in the ISO Document and the '484 patent

25 referenced above.

FIGURE 17 shows a tonal masker 1024 with its associated masking skirt 1028. The masking skirt 1028 represents a threshold indicating which signals will be masked by the tonal masker 1024. A signal that falls below the masking skirt 1028 (such as the signal designated 1030) cannot be heard because it is masked by the tone masker 1024. On the other hand, a smaller amplitude tone (such as tone 1032) can be heard because its amplitude rises above the masking skirt 1028.

As shown in FIGURE 17, the closer in frequency a signal is to the tonal masker 1024, the greater its amplitude may be and still be masked. Signals that have very different frequencies from the masker 1024, such as signal 1032, may have a lower amplitude and not fall below the masking skirt 1028, nor be masked.

Another aspect of hearing and psycho-acoustics is that a person can hear two tones at the same frequency provided that the softer tone is close enough in amplitude to the louder tone. The maximum difference in amplitude between the two tones of common frequency is referred to as the masking index. The masking index is dependent, in part, upon frequency of the tones. Generally, the masking index increases with frequency. For instance, the masking index of a masking tone at 1000 Hz will be smaller than the masking index of a masking tone at 7000 Hz.

FIGURE 17 illustrates the masking index 1034 for the tonal masker 1024. The masking index 1034 is the distance from the peak 1026 of the tonal masker 1024 to the top 1036 of the masking skirt

1028. This distance is measured in dB. For purposes of illustration, the graphs in FIGURES 16-19 scale the frequency along the modules of the graph in Bark. Each Bark corresponds to a frequency band distinguished by the human auditory system (also referred to as a "critical band"). The human ear divides the discernable frequency range into 24 critical bands. The frequency in psycho-acoustics is often measured in Bark instead of Hertz. There is a simple function that relates Bark to Hertz. The frequency range of 0 to 20,000 Hertz is mapped nonlinearly onto a range of approximately 0 to 24 Bark, according to a known function.

At low frequencies, the human ear/brain has the ability to discern small differences in the frequency of a signal if its frequency is changed. As the frequency of a signal is increased, the ability of the human ear to discern differences between two signals with different frequencies diminishes. At high frequencies, a signal must change by a large value before the human auditory system can discern the change.

As noted above, signals which lack a dominant frequency may be produce noise maskers. A noise masker is constructed by summing all of the audio energy within 1 Bark (a critical band) and forming a single discrete "noise" masker at the center of the critical band. Since there are 24 Bark (critical bands) then there are 24 noise maskers. The noise maskers are treated just like the tonal maskers. This means that they have a masking index and a masking

skirt. It is known that an audio signal may or may not have tonal maskers 1024, but it will generally have 1024 noise maskers.

FIGURE 18 illustrates a masking skirt 1029 similar to that described in the ISO/MPEG Layer-II for psycho-acoustic model I. The masking skirt 1029 is more complex than that of FIGURE 17. The masking skirt 1029 includes four mask portions 1050, 1052, 1054, and 1056, each of which has a different slope. The mask portions 1052-1056 are defined by the following equations:

- 1) Skirt Portion 1050 = E;
- 2) Skirt Portion 1052 =  $F * P + G$ ;
- 3) Skirt Portion 1054 = H; and
- 4) Skirt Portion 1056 =  $I - J * P$ ,

wherein the variables E, F, G, H, I and J represent psycho-acoustic parameters which are initially defined in preset tables, but may be adjusted by the user as explained below. The variable P represents the amplitude 1027 of the masker 1025 to which the masking skirt 1029 corresponds. Thus, the slopes of the mask portions 1050-1056 depend on the amplitude P of the masker 1025. The distance DZ, indicated by the number 1053, represents the distance from the masker 1025 to the signal being masked. As the distance DZ increased between the masker 1029 and the signal to be masked, the masker 1029 is only able to cover up lower and lower amplitude signals. The masking index, AV, indicated by the number 1055, is a function of the frequency. The masking index 1055 for tonal and noise maskers are calculated based on the following formula:

$$5) \quad AV_{\text{mask}} = A + B \cdot Z; \text{ and}$$

$$6) \quad AV_{\text{noise}} = C + D \cdot Z;$$

wherein the variables A, B, C and D represent psycho-acoustic parameters and the variable Z represents the frequency of the masker in Bark. The parameters A-J and suggested values therefor have been determined by readily available psycho-acoustic studies. A summary of such studies is contained in the book by Zweiker and Fastl entitled "Psychoacoustics," which is incorporated herein by reference.

#### 10 ISO/MPEG LAYER-II

The CODEC 1010 utilizes the psycho-acoustical model as described in the ISO psycho-acoustical model I as the basis for its parameters. The ISO model I has set standard values for ten model parameters (A, B, ... J). These model parameters are described

15 below:

	A	=	6.025 dB
	B	=	0.275 dB/Bark
	C	=	2.025 dB
	D	=	0.175 dB/Bark
20	E	=	17.0 dB/Bark
	F	=	0.4 1/Bark
	G	=	6.0 dB/Bark
	H	=	17.0 dB/Bark
	I	=	17.0 dB/Bark
25	J	=	.15 1/Bark

Parameters A through J are determined as follows:

$Z = \text{freq in Bark}$

$DZ = \text{distance in Bark from master peak (may be + or -) as shown in FIGURE 5}$

Pxx(Z(k)) = Power in SPL(96 db = +/-  
32767) at frequency Z of  
masker K

5                   xx       =       tm for tonal  
                              masker or nm  
                              for noise  
                              masker

Pxx is adjusted so that a full scale sine wave  
(+/-32767) generates a Pxx of 96 db.

10           Pxx = XFFT + 96.0 where XFFT = 0 db at +/-32767 amplitude

XFFT is the raw output of an FFT. It must be  
scaled to convert it to Pxx

AVtm(k) = A + B \* Z(k)   Masking index for tonal masker k

AVnm(k) = C + D \* Z(k)   Masking index for tonal masker k

15           VF(k,DZ) = E \* (|DZ| - 1) + (F \* X(Z(k))) + G)

VF(k,DZ) = (F \* X(Z(k))) + G \* |DZ|

VF(k,DZ) = H \* DZ

VF(k,DZ) = (DZ - 1) \* (I - J \* X(Z(k))) + H

MLxx(k,DZ) = Pxx(k) - (AVxx(K) + VF(k,DZ))

20           MLxx is the masking level generated by each masker k at  
a distance DZ from the masker.

where xx = tm or nm  
Pxx = Power for tm or nm

Parameters A through J are shown in FIGURE 15. Parameters A  
25   through J are fully described in the ISO 11172-3 document.

### Additional Parameters Added to ISO/MPEG LAYER-II

In addition to parameters A-J, the CODEC 1010 may use additional parameters K-Z and KK-NN. The CODEC 1010 allows the user to adjust all of the parameters A-Z and KK-NN. The additional parameters K-Z and KK-NN are defined as follows:

#### Parameter K - joint stereo sub-band minimum value

This parameter ranges from 1 to 31 and represents the minimum sub-band at which the joint stereo is permitted. The ISO specification allows joint stereo to begin at sub-band 4, 8, 12, or 16. Setting K to 5 would set the minimum to 8. Setting this parameter to 1 would set the minimum sub-band for joint stereo to 4.

#### Parameter L - anti-correlation joint stereo factor

This parameter attempts to determine if there is a sub-band in which the left and right channels have high levels, but when summed together to form mono, the resulting mono mix has very low levels. This occurs when the left and right signals are anti-correlated. If anti-correlation occurs in a sub-band, joint stereo which includes that sub-band cannot be used. In this case, the joint stereo boundary must be raised to a higher sub-band. This will result in greater quantization noise but without the annoyance of the anti-correlation artifact. A low value of L indicates that if there is a very slight amount of anti-correlation, then move the sub-band boundary for joint stereo to a higher value.

#### Parameter M - limit sub-bands

This parameter can range from 0 to 31 in steps of 1. It represents the minimum number of sub-bands which receive at least the minimum number of bits. Setting this to 8.3 would insure that sub-bands 0 through 7 would receive the minimum number of bits independent of the psychoacoustic model. It has been found that the psychoacoustic model sometimes determines that no bits are required for a sub-band and using no bits as the model specifies, results in annoying artifacts. This is because the next frame might require bits in the sub-band. This switching effect is very noticeable and annoying. See parameter { for another approach to solving the sub-band switching problem.

## Parameter N - demand / constant bit rate

5 This is a binary parameter. If it is above .499 then the demand bit rate bit allocation mode is requested. If it is below .499 then the fixed rate bit allocation is requested. If the demand bit rate mode is requested, then the demand bit rate is output and can be read by the computer. Also, see parameter R. Operating the CODEC in the demand bit rate mode forces the bits to be allocated exactly as the model requires. The resulting bit rate may be more or less than the number of bits available. 10 When demand bit rate is in effect, then parameter M has no meaning since all possible sub-bands are utilized and the required number of bits are allocated to use all of the sub-bands.

15 In the constant bit rate mode, the bits are allocated in such a manner that the specified bit rate is achieved. If the model requests less bits than are available, any extra bits are equally distributed to all sub-bands starting with the lower frequency sub-bands.

## 20 Parameter O - safety margin

This parameter ranges from -30 to +30 dB. It represents the safety margin added to the psychoacoustic model results. A positive safety margin means that more bits are used than the psychoacoustic model predicts, while a negative safety margin means to use less bits than the psychoacoustic model predicts. If the psychoacoustic model was exact, then this parameter would be set to 0. 25

## Parameter P - joint stereo scale factor mode

30 This parameter ranges from 0 to .999999. It is only used if joint stereo is required by the current frame. If joint stereo is not needed for the frame, then this parameter is not used. The parameter p is used in the following equation:

$$\text{br} = \text{demand bit rate} * p$$

35 If br is greater than the current bit rate (...128, 192, 256, 384), then the ISO method of selecting scale factors is used. The ISO method reduces temporal resolution and requires less bits. If br is less than the current bit rate, then a special method of choosing the scale factors is invoked. This special model generally requires that 40 more bits are used for the scale factors but it provides a better stereo image and temporal resolution. This is



generally better at bit rates of 192 and higher. Setting p to 0 always forces the ISO scale factor selection while setting p to .9999999 always forces the special joint stereo scale factor selection.

5      Parameter Q - joint stereo boundary adjustment

10      This parameter ranges from -7 to 7 and represents an adjustment to the sub-band where joint stereo starts. For example, if the psychoacoustic model chooses 14 for the start of the joint stereo and the Q parameter is set to -3, the joint boundary set to 11 (14 - 3). The joint bound must be 4, 8, 12 or 16 so the joint boundary is rounded to the closest value which is 12.

Parameter R - demand minimum factor

15      This value ranges from 0 to 1 and represents the minimum that the demand bit rate is allowed to be. For example, if the demand bit rate mode of bit allocation is used and the demand bit rate is set to a maximum of 256 kbs and the R parameter is set to .75 then the minimum bit rate is 192 kbs ( $256 * .75$ ). This parameter should not be  
20      necessary if the model was completely accurate. When tuning with the demand bit rate, this parameter should be set to .25 so that the minimum bit rate is a very low value.

Parameter S - stereo used sub-bands

25      This parameter ranges from 0 to 31 where 0 means use the default maximum (27 or 30) sub-bands as specified in the ISO specification when operating in the stereo and dual mono modes. If this parameter is set to 15, then only  
30      sub-bands 0 to 14 are allocated bits and sub-bands 15 and above have no bits allocated. Setting this parameter changes the frequency response of the CODEC. For example, if the sampling rate is 48,000 samples per second, then the sub-bands represent 750 HZ of bandwidth. If the used sub-bands is set to 20, then the frequency  
35      response of the CODEC would be from 20 to 15000 HZ ( $20 * 750$ ).

Parameter T - joint frame count

40      This parameter ranges from 0 to 24 and represents the minimum number of MUSICAM<sup>®</sup> frames (24 millisecond for 48k or 36 ms for 32k) that are coded using joint stereo. Setting this parameter non-zero keeps the model from switching quickly from joint stereo to dual mono. In the

ISO model, there are 4 joint stereo boundaries. These are at sub-band 4, 8, 12 and 16 (starting at 0). If the psychoacoustic model requires that the boundary for joint stereo be set at 4 for the current frame and the next frame can be coded as a dual mono frame, then the T parameter requires that the boundary be kept at 4 for the next T frames, then the joint boundary is set to 8 for the next T frames and so on. This prevents the model from switching out of joint stereo so quickly. If the current frame is coded as dual mono and the next frame requires joint stereo coding, then the next frame is immediately switched into joint stereo. The T parameter has no effect for entering joint stereo, it only controls the exit from joint stereo. This parameter attempts to reduce annoying artifacts which arise from the switching in and out of the joint stereo mode.

#### Parameter U - peak / rms selection

This is a binary parameter. If the value is less than .499, then the psychoacoustic model utilizes the peak value of the samples within each sub-band to determine the number of bits to allocate for that sub-band. If the parameter is greater than .499, then the RMS value of all the samples in the sub-band is used to determine how many bits are needed in each sub-band. Generally, utilizing the RMS value results in a lower demand bit rate and higher audio quality.

#### Parameter V - tonal masker addition

This parameter is a binary parameter. If it is below .499 the 3 db additional rule is used for tonals. If it is greater than .499, then the 6db rule for tonals is used. The addition rule specifies how to add masking level for two adjacent tonal maskers. There is some psychoacoustic evidence that the masking of two adjacent tonal maskers is greater (6db rule) than simply adding the sum of the power of each masking skirt (3db). In other words, the masking is not the sum of the powers of each of the maskers. The masking ability of two closely spaced tonal maskers is greater than the sum of the power of each of the individual maskers at the specified frequency. See FIGURE 6.

#### Parameter W - sub-band 3 adjustment

This parameter ranges from 0 to 15 db and represents an adjustment which is made to the psychoacoustic model for sub-band 3. It tells the psychoacoustic model to

5 allocate more bits than calculated for this sub-band. A value of 7 would mean that 7db more bits (remember that 1 bit equals 6 db) would be allocated to each sample in sub-band 3. This is used to compensate for inaccuracies in the psychoacoustic model at the frequency of sub-band 3 (3\*750 to 4\*750 Hz for 48k sampling).

Parameter X - adj sub-band 2 adjustment

10 This parameter is identical to parameter W with the exception that the reference to sub-band 3 in the above-description for parameter W is changed to sub-band 2 for parameter X.

Parameter Y - adj sub-band 1 adjustment

15 This parameter is identical to parameter W with the exception that the reference to sub-band 3 in the above-description for parameter W is changed to sub-band 1 for parameter Y.

Parameter Z - adj sub-band 0 adjustment

20 This parameter is identical to parameter W with the exception that the reference to sub-band 3 in the above-description for parameter W is changed to sub-band 0 for parameter Z.

Parameter KK - sb hang time

5 The psychoacoustic model may state that at the current time, a sub-band does not need any bits. The KK parameter controls this condition. If the parameter is set to 10, then if the model calculates that no bits are needed for a certain sub-band, 10 consecutive frames must occur with no request for bits in that sub-band before no bits are allocated to the sub-band. There are 32  
30 counters, one for each sub-band. The KK parameter is the same for each sub-band. If a sub-band is turned off, and the next frame needs bits, the sub-band is immediately turned on. This parameter is used to prevent annoying switching on and off of sub-bands. Setting this  
35 parameter non-zero results in better sounding audio at higher bit rates but always requires more bits. Thus, at lower bit rates, the increased usage of bits may result in other artifacts.

40 Parameter LL - joint stereo scale factor adjustment

5 If this parameter is less than .49999, then scale factor adjustments are made. If this parameter is .5000 or greater, then no scale factor adjustments are made (this is the ISO mode). This parameter is used only if joint stereo is used. The scale factor adjustment considers the left and right scale factors a pair and tries to pick a scale factor pair so that the stereo image is better positioned in the left/right scale factor plane. The result of using scale factor adjustment is that the stereo image is significantly better in the joint stereo mode.

Parameter MM - mono used sub-bands

This parameter is identical to parameter S except it applies to mono audio frames.

15 Parameter NN - joint stereo used sub-bands

This parameter is identical to parameter S except it applies to joint stereo audio frames.

As the psycho-acoustic parameters affect the resultant quality of the audio output, it would be advantageous for users to vary the output according to the user's desires.

20 In a preferred embodiment of the disclosed CODEC 1010, the psycho-acoustic parameters can be adjusted by the user through a process called dynamic psycho-acoustic parameter adjustment (DPPA) or tuning. The software for executing DPPA is disclosed in the incorporated Software Appendix and discussed in more detail below in connection with Figs. 21-32. DPPA offers at least three important advantages to a user of the disclosed CODEC over prior art CODECs. First, DPPA provides definitions of the controllable parameters and their effect on the resulting coding and compression processes. Second, the user has control over the settings of the defined DPPA parameters in real time. Third, the user can hear the

result of experimental changes in the DPPA parameters. This feedback allows the user to intelligently choose between parameter alternatives.

5        Tuning the model parameters is best done when the demand bit rate is used. Demand bit rate is the bit rate calculated by the psycho-acoustic model. The demand bit rate is in contrast to a fixed bit rate. If a transmission facility is used to transmit compressed digital audio signals, then it will have a constant bit rate such as 64, 128, 192, 256 ... kbs. When tuning the parameters  
10        while using the Parameter N described above, it is important that the demand bit rate is observed and monitored. The model parameters should be adjusted for the best sound with the minimum demand bit rate. Once the parameters have been optimized in the demand bit rate mode, they can be confirmed by running in the  
15        constant bit rate mode (see Parameter N).

      DPPA also provides a way for the user to evaluate the effect of parameter changes. This is most typically embodied in the ability for the user to hear the output of the coding technique as changes are made to the psycho-acoustic parameters. The user can  
20        adjust a parameter and then listen to the resulting change in the audio quality. An alternate embodiment may incorporate measurement equipment in the CODEC so that the user would have an objective measurement of the effect of parameter adjustment on the resulting audio. Other advantages of the disclosed invention with the DPPA  
25        are that the user is aware of what effect the individual parameters

have on the compression decompression scheme, is able to change the values of parameters, and is able to immediately assess the resulting effect of the current parameter set.

5 One advantage of the ability to change parameters in the disclosed CODEC, is that the changes can be accepted in real time. In other words, the user has the ability to change parameters while the audio is being processed by the system.

10 In the preferred embodiment, the compression scheme (attached as the Software Appendix) includes thirty adjustable parameters. It is contemplated that additional parameters can be added to the CODEC to modify the audio output. Provisions have been made in the CODEC for these additional parameters.

15 Turning now to FIGURE 19, one can see two tonal maskers 1024 and 1025. The individual masking skirts for these maskers are shown in 1028. The encoder predicts how do these individual maskers mask a signal in the region in between 1024 and 1025. The summing of the masking effects of each of the individual maskers may be varied between two methods of summing the effects of tonal maskers. These methods are controlled by Parameter V described  
20 above.

FIGURE 20 is illustrative of the steps the user must take to modify each parameter. As shown in FIGURE 20, the parameters are set to their default value (which may be obtained from one of several stored table) and remain at that value until the user  
25 adjusts the parameter. The user may change the parameter by

turning one of the knobs, pushing one key on the keypad, or changing one of the graphics representative of one of the parameters on the computer monitor. Thus, as shown in box 1060, the disclosed CODEC 1010 waits until the user enters a command directed to one of the parameters. The CODEC 1010 then determines which parameter had been adjusted. For example, in box 1062 the CODEC inquires whether the parameter that was modified was parameter J. If parameter J was not selected, the CODEC 1010 then returns to box 1060 and awaits another command from the user. If parameter J was selected, the CODEC 1010 awaits for the user to enter a value for that parameter in box 1064. Once the user has entered a value for that parameter, the CODEC 1010, in box 1066, stores that new value for parameter J. The values for the default parameters are stored on a storage medium in the encoder 1012, such as a ROM or other chip.

Turning again to FIGURES 14 and 15 (which generally illustrate the operation of the disclosed CODEC) an analog audio source 1016 is fed into the encoder/decoder (CODEC) 1010 which works in loop back mode (where the encoder directly feeds the decoder). Parametric adjustments can be made via a personal computer 1040 attached to the CODEC 1010 from an RS232 port (not shown) attached to the rear of the CODEC. A cable 1042 which plugs into the RS232 port, connects into a spare port (not shown) on the PC 1040 as shown in FIGURE 14. The personal computer 1040 is preferably an IBM-PC or IBM-PC clone, but can be an any personal computer

including a Mackintosh\*. The personal computer 1040 should be at least a 386DX-33, but is preferably a 486. The PC should have a VGA monitor or the like. The preferred personal computer 1040 should have at least 4 mb of memory, a serial com port, a mouse, and a hard drive.

Once the PC 1040 is connected to the CODEC 1010, a tuning file can be loaded onto the personal computer 1040, and then the parameters can be sent to the encoder via a cable 1042. A speaker 1044 is preferably attached to the output of the CODEC 1010, via a cable 1046, to give the user real time output. As a result, the user can evaluate the results of the parameter adjustment. A headphone jack (not shown) is also preferably included so that a user can connect headphones to the CODEC and monitor the audio output.

The parameters can be adjusted and evaluated in a variety of different ways. In the preferred embodiment, a mouse is used to move a cursor to the parameter that the user wishes to adjust. The user then holds down the left mouse button and drags the fader button to the left or right to adjust the parameter while listening to the audio from the speaker 1044. For example, if the user were to move the fader button for parameter J to the extreme right, the resulting audio would be degraded. With this knowledge of the system, parameter J can be moved to test the system to insure that the tuning program is communicating with the encoder. Once the



user has changed all or some of the parameters, the newly adjusted parameters can be saved.

In another embodiment, control knobs or a keypad (not shown), can be located on the face of the CODEC 1010 to allow the user to adjust the parameters. The knobs would communicate with the tuning program to effectuate the same result as with the fader buttons on the computer monitor. The attachment of the knobs can be hard with one knob allotted to each adjustable parameter, or it could be soft with a single knob shared between multiple parameters.

10 In another embodiment, a graphic representing an "n" dimensional space with the dimensions determined by the parameters could be shown on the computer display. The operator would move a pointer in that space. This would enable several parameters to be adjusted simultaneously. In still another embodiment, the parameters can be adjusted in groups. Often psycho-acoustic parameters only make sense when modified in groups with certain parameters having fixed relationships with other parameters. These groups of parameters are referred to as smart groups. Smart group adjustment would mean that logic in the CODEC would change related parameters (in the same group) when the user changes a given parameter. This would represent an acceptable surface in the adjustable parameter space.

25 In yet another embodiment, a digital parameter read out may be provided. This would allow the values of the parameters to be digitally displayed on either the CODEC 1010 or the PC 1040. The

current state of the CODEC 1010 can then be represented as a simple vector of numbers. This would enable the communication of parameter settings to other users.

Parameter adjustment can be evaluated in ways other than by  
5 listening to the output of speaker 1044. In one embodiment, the CODEC 1010 is provided with an integrated FFT analyzer and display, such as shown in applicant's invention entitled "System For Compression And Decompression Of Audio Signals For Digital Transmission," and the Software Appendix that is attached thereto,  
10 that are both hereby incorporated by reference. By attaching the FFT to the output of the CODEC, the user is able to observe the effect of parametric changes on frequency response. By attaching the FFT to the input of the CODEC, the user is able to observe frequency response input. The user can thus compare the input  
15 frequency response to the output frequency response. In another embodiment, the disclosed CODEC 1010 is provided with test signals built into the system to illustrate the effect of different parameter adjustments.

In another embodiment, the DPPA system may be a "teaching  
20 unit." To determine the proper setting of each parameter, once the determination is made, then the teacher could be used to disburse the parameters to remote CODECs (receivers) connected to it. Using this embodiment, the data stream produced by the teaching unit is sent to the remote CODEC that would then use the data stream to  
25 synchronize their own parameters with those determined to be

appropriate to the teacher. This entire system thus tracks a single lead CODEC and avoids the necessity of adjusting the parameters of all other CODECs in the network of CODECs.

#### Processing Flow of the Preferred Embodiment

5       Next, the processing flow of the preferred embodiment is described in connection with Figs. 21-33.

Fig. 21 generally illustrates the functions of an encoder for a single channel receiving audio signal. The encoder includes a plurality of band pass filters separately divided into a low pass filter bank 502 and a high pass filter bank 504. The low and high pass filter banks 502 and 504 include a plurality of band pass filters 506. The number of band pass filters in each filter bank may be dynamically varied during joint stereo framing by the psycho-acoustic processor as explained below. For purposes of illustration, four filters have been dynamically assigned to the low pass filter bank 502, and the remaining filters have been assigned to the high pass filter bank 504. The band pass filters 506 receive a segment of predefined length (e.g., 24ms) of an incoming analog audio signal and pass corresponding subbands thereof. Each band pass filter 506 is assigned to a separate pass band having a unique center frequency and a corresponding bandwidth. The widths of each pass band may differ, for instance, whereby the band pass filters for low frequency signals have narrower pass bands than the pass bands of filters corresponding to

high frequency signals. The band pass filters are defined such that the pass bands slightly overlap.

The subband signals output by the band pass filters 506 are delivered to corresponding scalers 508 which adjust the gain of the subband signals and deliver same to corresponding quantizers 510. The subband signals received by each scaler 508 are divided into a predetermined number of blocks (e.g. three blocks each of which is 8 milliseconds in length for a 24 millisecond segment of audio data). The scalers 508 adjust the gain of the corresponding subband signal for each block within a segment until the peak to peak amplitude of the subband signal substantially corresponds to the range of the quantizer 510. The gain of the subband signal is controlled by the scaler 508 to ensure that the peak to peak amplitude never exceeds the capacity of the quantizer 510. By way of example, each subband signal delivered from a band pass filter 506 may include 36 samples divided into three blocks of 12 samples. The scaler 508 adjusts the gain of the 12 sample blocks as explained above to ensure that the quantizer 510 is fully loaded. The quantizer 510 has a maximum quantization capacity. The quantizers 510 convert the incoming samples to one of a predefined number of discrete levels and outputs a corresponding digital signal representative of the closest quantization level to the sample level. The number and distance between quantization levels is governed by the number of bits allocated to the quantizer 510. For instance, the quantizer 510 will use more quantization levels

if afforded 10 bits per sample as compared to the number of quantization levels which correspond to 6 bits per sample. As more bits are assigned to the quantizer, the sample is more accurately digitized and less noise is introduced. The quantizers 510 deliver  
5 output quantized subband signals to a multiplexer 512, which combines the subband signals to form a frame of data which is ultimately transmitted by the encoder.

A psycho-acoustic processor (PAP) 514 process the incoming analog audio signal (as explained below) and controls the  
10 quantizers 510 and scalers 508 to allocate the minimum necessary number of bits to each quantizer. In accordance with the process explained below, the PAP 514 may direct the quantizer 516 to utilize six bits per sample, while limiting quantizer 518 to two bits per sample.

15 Fig. 22 generally illustrates a frame 530 having a header segment 532, a data segment 534, and an ancillary data segment 536. The data segment 534 includes multiple subband components 538, each of which corresponds to a unique subband ( $SB_1$ - $SB_{j_2}$ ). Each subband component 538 is divided into three blocks 540, each of which has  
20 been scaled by the scaler 508 to properly load the quantizer 510. It is to be understood that the blocks 540 and subband components 538 will vary in length depending upon the number of bits used by the corresponding quantizer 510 to encode the corresponding subband signal. For instance, when quantizer 516 is directed (by the path  
25 514) to use six bits per sample, the corresponding data component

542 will include 18 bits of data (six bits per block). However, when quantizer 518 is assigned two bits per sample, data component 544 will include six bits (two bits per block). The audio data segment 534 has a fixed maximum length, and thus a limited number of bits are available for use by the quantizers 510. The PAP 514 maximizes the bit allocation between the quantizers 510.

Once the bit allocation is complete, the PAP 514 loads the corresponding subsection and the header segment 532 with the corresponding encoder information 546. The encoder information 546 includes the number of bits allocated to each quantizer 510 for the corresponding subband (referred to hereafter as the "Bit Allocation Information 548"). The encoder information 546 further includes the scaling factors 550 used by the scalars 508 in connection with corresponding blocks 540 of corresponding subband components 538. In addition, the encoder information 546 includes scaling factor sample information 552 (explained below).

Fig. 23 illustrates an encoder including the structure of the encoder from Fig. 21, with the further ability to offer joint stereo at a decoder output. In Fig. 23, the encoder is generally denoted by block 600, and the decoder is denoted by block 602. The encoder 600 receives a stereo signal upon left and right channels. The decoder 602 outputs a joint stereo signal at speakers 604 and 606. The encoder 600 includes low pass filter banks (LPFB) 608 and 612 corresponding to the left and right channels, respectively. The encoder 600 further includes high pass filter banks (HPFB) 610

and 614, also corresponding to the left and right channels, respectively. The low and high pass filter banks 608-614 include a plurality of band pass filters which are controlled by a PAP, as explained in connection with Fig. 21. The output signals of the  
5 low pass filter banks 608 and 612 are delivered to scaler banks 616 and 618, each of which also include a plurality of scalers which operate in a manner similar to the scalers 508 in Fig. 21. The scaler banks 616 and 618 deliver scaled signals to quantizer banks 620 and 622, each of which similarly includes a plurality of  
10 quantizers similar to quantizers 510 in Fig. 21.

While not showing, it is understood that the filter banks 616 and 618 and the quantizers 620 and 622 controlled by a PAP similar to the psycho-acoustic processor 514 in Fig. 21. The low pass filter banks 608 and 612, scaler banks 616 and 618, and quantizer  
15 banks 620 and 622 cooperate to separately encode the lower subbands for the left and right channels of the stereo input signal. The encoded signals for the lower subbands are in turn delivered from the quantizers 620 and 622 and ultimately received by corresponding inverting quantizers 624 and 626. The inverting quantizers 624 and  
20 626 cooperate with inverse scaling banks 628 and 630 to reconvert the lower frequency portions of the encoded left and right channel signals back to analog audio.

The encoder 600 further includes a summer 632 which combines the output signals from the high pass filter banks 610 and 614 for  
25 the left and right channels to produce a joint mono signal for the

higher pass bands. The output of the summer 632 is in turn delivered to a scaling bank 634, which scales the signal to properly load the quantizer bank 636. The output signal of the quantizer bank 636 is delivered to an inverse quantizer 638 to  
5 reverse the process. The output of the inverse quantizer 638 is delivered to two scaling banks 640 and 642 which are controlled via control channels 644 and 646.

The encoder 600 further includes calculating modules 650 and 652, which measure the energy in the corresponding high pass  
10 subbands. The modules 650 and 652 then adjust the gain of scalers 640 and 642 in proportion to the energy of the corresponding high pass subbands. For instance, if HPFB 610 outputs more energy than HPFB 614, then scaler 640 is set to boost the gain of its input signal greater than the gain boost of scaler 642. Thus, the audio  
15 signal in the higher pass bands is output predominantly at speaker 604. The energy calculator 650 and 652 may be carried out by the psycho-acoustic processor in a manner explained below.

Next, the discussion turns to the process followed by the present invention to undergo encoding.

20 With reference to Fig. 24, the PAP 514 cooperates with the quantizer 510 and scaler 508 to digitize the analog audio signals received from each band pass filter 506 for corresponding subbands (step 2400). In step 2402, the digitized signals for the subbands from each bandpass filter are divided into a predefined number of  
25 blocks. For example, a 24 millisecond segment of analog audio may



be converted to 36 digital samples and then divided into three blocks of 12 samples each. In step 2404, each block of samples is analyzed to determine the maximum amplitude of the digitized signal therein. In step 2406, the scalers 508 are adjusted to vary the scale of the samples within each block until the samples correspond to a signal gain substantially equalling the range of the quantizers 510.

Turning to Figs. 25A and 25B, while the scalers 508 are being adjusted (as explained in connection with Fig. 24), the PAP 514 calculates the global masking threshold (GMT) to be used in connection with the present sample of analog audio data. Beginning at step 2502, the PAP 514 obtains a working table of psycho-acoustic parameters having a value for each of parameters A-NN (described above). The table of parameters may be one of several predefined tables stored in memory in the encoder. The table is updated dynamically by the user during operation of the encoder. For instance, when the encoder is initially started, an initial set of parameter values may be read from the encoder memory and used to initialize the encoder. Thereafter, as the PAP 514 continuously processes segments of analog audio data, the user may vary the parameter values stored in the working table. Once the user varies a parameter value in the working table, the PAP 514 obtains the new parameter value set for processing the following analog audio segments. For instance, after the user listens to a short segment (one minute) of analog audio encoded and decoded according to the

initial working table, the user may desire to adjust the parameters within the working table. Once the user adjusts these parameters, the PAP 514 effects subsequent psycho-acoustic processing based on the new parameter values assigned by the user. Thus, the user is  
5 afforded the opportunity to listen to the signal which results from the users adjustment in the parameters.

Returning to Fig. 25A, once the PAP 514 obtains the working table of parameters A-NN, the PAP 514 uses these parameter values for the current segment of audio data. At step 2504, the PAP 514  
10 obtains a segment of analog audio data of predetermined length (e.g., 24 milliseconds). The segment is digitized. At step 2506, the PAP 514 converts the digitized segment from the time to the frequency domain according to the bark scale. These conversions may be effected using a Fast Fourier Transform and a known Bark  
15 transfer function between the bark frequency domain and the normal frequency domain. At step 2508, the PAP calculates the threshold of hearing. At step 2510, the PAP analyzes the signal converted in step 2506 to the bark frequency domain to locate the tonal peaks therein. Once located, the tonal peaks are removed in step 2512  
20 from the digital converted signal. Next, the digitized signal is divided into critical bands (step 2514). Noise maskers are calculated for each critical band by summing the remaining energy within each critical band (after the tonal peaks have been removed). A representative noise masker is obtained for each  
25 critical band from the noise calculated in step 2514. It is

understood that, a signal noise masker is substituted therefore at a single frequency and having a predetermined amplitude. The amplitude and frequency of the noise masker are determined by the amount of noise energy within the critical band.

5       At step 2516 (Fig. 25B), the PAP calculates masking skirts for the tonal and noise maskers based on parameters A-J and based on the amplitudes and frequencies of the tonal and noise maskers. At step 2518, the PAP combines the noise and tonal masking skirts and the threshold of hearing to obtain a global masking threshold for  
10       the presently digitized segment of audio data. The global masking threshold (GMT) is divided into subbands at step 2520. The subbands correspond to the band pass filters 506. At step 2520 the PAP locates the maximum and minimum of each global masking threshold within each subband. At step 2522 the PAP assigns  
15       quantization levels for each subband based on amount of noise which may be added to each subband without exceeding the minimum value of the GMT within the corresponding subband. The assignment process is described in more detail below.

Turning to Fig. 26, the process of obtaining the GMT is  
20       explained in more detail. At step 2600, the PAP locates the first subband (subband 0) and obtains the first masker within this subband (step 2602). At step 2604, the PAP combines the current masker obtained in step 2602 with the threshold of hearing to obtain an initial GMT for the subband. Thereafter the next masker  
25       is obtained at step 2606. The PAP then determines at step 2608

whether the newly obtained and preceding maskers represent adjacent tonal maskers. If two adjacent tonal maskers are being combined, control flows to step 2610 at which the PAP combines the two adjacent total maskers within the GMT using one of two addition rules defined by parameter V. For instance, the two tonal maskers may be combined according to a 3db or a 6db addition rule based upon which is chosen by the parameter V. The tonal maskers are combined according to one of the following equations:

$$3db(rule) = 10 \log_{12}(10 P_{1(db)} \backslash 10 + 10 P_{2(db)} \backslash 10)$$

$$6db(rule) = 2 \log_{12}(1 P_{1(db)} \backslash 2 + 1 P_{2(db)} \backslash 2)$$

Returning to step 2608 if the two maskers are not tonal maskers, flow moves to step 2612 at which the maskers are combined with the global masking threshold according to the conventional method. Next, at step 2614 it is determined whether the current masker represents the last masker in the subband. If not, steps 2606-2612 are repeated. If the current masker represents the last masker in the subband, flow passes to step 2616 at which the PAP determines whether the current subband is one of subbands 0, 1, 2 and 3. If so, control passes to step 2618 at which the global masking threshold for the current subband is adjusted by a biasing level determined by the corresponding one of parameter W-Z. For instance, if the current subband is subband 2, then the GMT within subband 2 is adjusted by a db level determined by parameter Y. At step 2620 it is determined whether the last subband has been analyzed. If not, flow pass to step 2602 where the above described

processes repeated. Otherwise, control returns to the main routine illustrated in Fig. 25.

Next, the quantization level assignment process of step 2522 is described in more detail in connection with Fig. 30. The assignment process involves three primary phases, namely an allocation phase, a deallocation phase and an excess bit allocation phase. During the allocation phase step (3000), the PAP steps through each subband for each channel (left and right) and assigns the corresponding quantizer a number of bits to be used for quantizing the subband signal. During bit allocation, the number of bits allocated to a subband are incremented in predefined allocation steps until a sufficient number of bits are assigned to the quantizer to ensure that the noise introduced into the signal during the quantizing process is below the minimum of the GMT for the subband. Once the necessary number of bits are assigned to each subband at step 3000 it is determined whether the number of bits allocated has exceeded the number of bits available (i.e., the bit pool) at step 3002. If not, and extra bits exist then control flows to step 3004. At step 3004, the PAP determines whether the encoder is operating in a demand or constant bit rate mode. In a demand mode, once the PAP allocates bits to each subband, the allocations become final, even through the total number of bits needed is less than the number available for the current transmission rate. Thus, the allocation routine ends. However,

when in a constant bit rate mode, the extra bits are distributed evenly or unevenly among the subbands.

It is desirable to choose the demand bit rate made when tuning the codec to ensure that the signal heard by the user accurately reflects the parameter values set by the user. The remaining bits from the bit pool are distributed amongst the subbands to further reduce the quantization noise. However, if bit allocation in step 3000 has exceeded the bit pool then flow passes to step 3006 at which bit deallocation is performed and previously assigned bits are removed from selected quantizers which are deemed the best candidate for deallocation. Deallocation occurs with respect to those subbands at which deallocation will have the least negative effect. Put another way, the PAP deallocates bits from subbands which will continue, even after deallocation, to have quantization noise levels closest to the GMT minimum for that subband (even though the quantization noise level exceeds the GMT minimum).

During bit allocation, flow passes at step 3000 to the routine illustrated in Fig. 27. At step 2702, the PAP determines whether the encoder is operating in a stereo, mono, or joint stereo framing mode. The PAP sets the last subband to be used which is determined by the subband limit parameters S, MN and NN. At step 2704, the PAP determines the total number of bits available (i.e., the bit pool) for the current framing mode, namely for joint stereo, stereo or mono. At step 2706, the first subband and first channel are obtained. At step 2708, the maximum for the signal within the

current subband is compared to the GMT minimum within the current subband. If the subband signal maximum is less than the GMT minimum, then the current subband signal need not necessarily be transmitted since it falls below the GMT. Thus, flow passes to

5 step 2710 at which it is determined whether the current subband falls below a subband limit (defined by parameter M). If the current subband is below the subband limit then the PAP allocates bits to the subband even through the subband signal falls below the GMT minimum. For instance, if the current subband is two and the

10 user has designated (via parameter M) that subbands 0-5 should be encoded and transmitted, then subband 2 would be encoded by the corresponding quantizer with a minimum number of bits allocated to the quantizer. Thus, at step 2710, if the current subband is less than the subband limit then control passes to step 2712 at which

15 the bit allocation routine is called to assign at least a first allocation step of a minimum number of bits to the current subband. However, at step 2710 if it is determined that the current subband is greater than the subband limit then control passes to step 2718 and the bit allocation routine is bypassed (i.e. the quantizer for

20 the current subband is not assigned any bits and thus the signal within the current subband is not encoded, nor transmitted). At step 2712, prior to performing the bit allocation routine, the digitized audio signal within the current subband is adjusted to introduce a safety margin or bias thereto to shift the digitized

signal upward or downward. This safety margin represents a parameter adjusted dynamically by the user (parameter O).

After flow returns from the bit allocation routine, it is determined at step 2714 whether the encoder is operating in a joint stereo mode. If not flow passes to step 2718 at which it is determined whether the foregoing process (steps 2708-2714) need to be repeated for the opposite channel. If so, the channels are switched at step 2724 and the process is repeated. If not, flow passes from step 2718 to 2722 at which it is determined whether the current subband is the last subband. If not, the current subband is incremented at step 2726 and the allocation routine is repeated. Thus, steps 2708-2726 are repeated each subband.

Returning to step 2714, when operating in a joint stereo mode, control passes to step 2716 at which it is determined whether the bit allocation routine at step 2712 allocated a number of bits to the current subband which resulted in the total number of allocated bits exceeding the available bit pool for the current mode. If so, the current subband number is recorded at step 2720 as the subband at which the bit pool boundary was exceeded.

When in a stereo mode the process flows from step 2708 to step 2726 without using steps 2716 and 2720 in order that every subband within the right and left channels is assigned the necessary number of bits to insure that the quantization noise falls below the global masking threshold within the corresponding subband. When in the joint stereo mode, the foregoing process is repeated separately



for every subband within the left and right channels (just as in the stereo mode). However, the system records the subband number at which the available bit pool was exceeded in step 2720. This subband number is later used to determine a joint stereo boundary such that all subbands below the boundary are processed separately in stereo for the left and right channels. All subbands above the boundary are processed jointly, such as shown by the joint stereo encoder of Fig. 23. The subband boundary corresponds to the break point between the low pass filter banks 608 and 612 and the high pass filter banks 610 and 614 (shown in Fig. 23).

Turning to Fig. 28, the bit allocation routine is described in more detail. Beginning at step 2802, an array of allocation steps is obtained for the current mode (e.g., stereo, mono or joint stereo). Each level within the array corresponds to a predefined number of bits to be assigned to a quantizer. By way of example, the array may include 17 elements, with elements 1, 2 and 3 equaling 60 bits, 84 bits and 124 bits, respectively. Thus, at the first step 60 bits are assigned to the quantizer corresponding to the current subband. At the second step, 84 bits are assigned to the quantizer corresponding to the current subband. Similarly, at the third step, 124 bits are assigned to the quantizer for the current subband. The steps are incremented until the current step allocates a sufficient number of bits to the quantizer to reduce the quantization noise below the minimum GMT for the current subband. In addition to the bit allocation array, a mask to noise

ratio array is included containing a list of elements, each of which corresponds to a unique step. Each element contains a predefined mask to noise ratio identifying the amount of noise introduced into the encoded signal when a given number of bits are utilized to quantize the subband. For instance, steps 1, 2 and 3 may correspond to mask to noise ratios (MNR) of 10db, 8db and 6db, respectively. Thus, if 60 bits are allocated to the current quantizer for quantizing the current subband, 10db of noise will be introduced into the resultant encoded signals. Similarly, if 84 bits are used to quantize the signal within the current subband, 8db of noise are introduced.

At step 2802, the allocation and MNR arrays are obtained and the current step is set to 1. At step 2804, the allocation array is accessed to obtain the number of bits to be allocated to the current subband for the current step. At step 2806 the maximum level of the audio signal within the current subband is obtained based on one of the audio peak or RMS value, which one selected between determined by parameter U. Next, the MNR value for the current step is obtained from the MNR array (2808). At step 2810, it is determined whether the audio signal maximum, when combined with the MNR value of the current allocation step, exceed the minimum of the GMT for the current subband. If so, then a detectable amount of noise will be introduced into the signal if the current allocation step is used. Thus, control passes to step 2816.

At step 2816, the PAP records the difference between the GMT minimum of the current subband and the level combined signal formed from the maximum value for the audio signal and the MNR. Thereafter, at 2818 the allocation step is incremented in order to  
5 allocation more bits to the current subband. The foregoing loop is repeated until the allocation step is incremented sufficiently to allocate a number of bits to the current subband necessary to reduce the combined signal formed from the audio signal max and MNR below the minimum of the GMT. Once it is determined at step 2810  
10 that this combined signal is less than the minimum of the GMT, control passes to step 2812. At step 2812, the number of bits corresponding to the current step are allocated to the quantizer for the current subband. At step 2814, the system updates the total number of allocated bits for the current segment of audio  
15 information.

According to foregoing process, each quantizer is assigned a number of bits corresponding to an allocation step which is just sufficient to reduce the combined noise and audio signal below the minimum of the GMT. In addition, at step 2816, the system retains  
20 a deallocation table having one element for each subband and channel. Each element within the table corresponds to the difference between the GMT minimum and the combined audio signal maximum and MNR value for the allocation step preceding the allocation step ultimately assigned to the quantizer in step 2812.

By way of example, a quantizer may be assigned the number of bits corresponding to allocation step 3 (e.g., 124 bits). At step 2816, it was determined that the signal and MNR for step 2 exceeded the GMT minimum by 3db. The deallocation table will record at step 2816 this 3db value indicating that, while the current quantizer is assigned to allocation step 3, if the current quantizer had been assigned to allocation step #2, the combined signal and MNR would exceed the GMT minimum by 3db. The deallocation table recorded at step 2816 may be used later if the deallocation of bits becomes necessary (as explained below).

The bit allocation routine of Fig. 28 is continuously repeated for each channel and for each subband (according to the process of Fig. 27). Once control returns to step 3000 in Fig. 30, all of the subbands for both channels have been allocated the necessary number of bits. At step 3002 if it is determined that the number of bits allocated exceeds the bit pool, control passes to step 3006 which is illustrated in more detail in Fig. 31.

When it is determined that deallocation is necessary, control passes from step 3006 (Fig. 30) to the deallocation routine illustrated in Fig. 31. At step 3102, it is determined whether the encoder is operating in a joint stereo mode. If so, control passes to step 3104 at which the joint stereo boundary is determined. The joint stereo boundary represents the boundary between the low pass filter banks 608 and 612 and high pass filter banks 610 and 614 (Fig. 23). Subbands below the joint stereo boundary are processed

separately for the left and right channels within the low pass filter banks 608 and 612. Subbands above the joint stereo boundary are included within the high pass filter banks 610 and 614 and are combined in summer 632 to form a mono signal. Thus, subbands above  
5 the joint stereo boundary are combined for the left and right channels and passed through a single quantizer bank 636.

Returning to Fig. 31, once the joint stereo boundary is determined, a new bit pool is obtained based on the joint stereo boundary (step 3106). A new bit pool must be calculated since the  
10 original bit pool which calculated based on full stereo whereby it was presumed that bits would be allocated to all of the subbands separately for the left and right channels. However, subbands above the boundary are combined for the left and right channels and thus additional bits are available for allocation. For instance,  
15 in a full stereo system using 22 subbands per channel, bits must be allocated between 44 separate subbands (i.e., 22 subbands for the left channel and 22 subbands for the right channel). However, in a joint stereo mode utilizing 22 subbands with the joint stereo boundary at subband 8, only 32 subbands are necessary (i.e., eight  
20 lower subbands for the left channel, eight lower subbands for the right channel and 16 upper subbands for the combined signals from the left and right signals). Once the new bit pool is calculated, the joint stereo array is obtained at step 3108. The joint stereo array identifies the allocation steps combining the number of bits  
25 to be allocated for each step during the bit allocation routine

(Fig. 28). In addition, the joint stereo array identifies the mask to noise ratio for each allocation step. At step 3110, the bit allocation routine (Fig. 28) is called to allocate bits to the subbands, wherein subbands below the joint stereo boundary are separately allocated for the left and right channels, while  
5 subbands above the joint stereo boundary are allocated for a single set of band pass filters representing the combination of the signals from the left and right channels.

Next, at step 3112, it is determined whether the bit  
10 allocation for the joint stereo frame exceeds the joint stereo bit pool (obtained at step 3106). If not, control returns to the routine in Fig. 30. However, if more bits have been allocated than are available in the bit pool, control passes to step 3114 to begin a deallocation process. At step 3114, the deallocation table  
15 (generated at step 2816 in Fig. 28) is sorted based on the difference values recorded therein to align these difference values in descending order. At step 3116, the first element within the deallocation table is obtained. At step 3118, a deallocation operation is effected. To deallocate bits, the quantizer  
20 corresponding to the channel and subband identified in the first element of the deallocation table is assigned a new number of quantizing bits. The number of bits newly assigned to this quantizer corresponds to the step preceding the step original assigned to the quantizer. For instance, if during the original  
25 allocation routine, a quantizer was assigned 124 bits

(corresponding to step 3), then at step 3118, the quantizer would be assigned 84 bits (corresponding to allocation step 2).

At step 3120, a new difference value is calculated for the current subband based on the allocation step preceding the newly assigned allocation step. This new difference is added to the difference table at step 3122. The number of deallocated are then subtracted from the allocated bit total (step 3124). Thereafter, it is determined whether the new total of bits allocated still exceeds the available bit pool (step 3126). If not, control returns to step 3006 (Fig. 30). If the allocation bit total still exceeds the bit pool, control returns to step 3114 and the above described deallocation processes is repeated.

Figs. 32 and 33 set forth an example explained hereafter in connection with the allocation steps and deallocation routine. Figs. 32A and 32B illustrate two exemplary subbands with the corresponding portions of the global masking threshold and the quantized signal levels derived from the audio signal peak and MNR value. The quantized signal levels are denoted at points 3106-3108 and 3110-3113. The minimums of the GMT are denoted at levels 3204 and 3205. Stated another way, if the number of bits associated with allocation step #1 are assigned to the quantizer for subband 3 (Fig. 32A), the resultant combined audio signal and MNR will have a magnitude proximate line 3206. If more bits are assigned to the quantizer (i.e., allocation step #2), the combined signal and MNR value is reduced to the level denoted at line 3207. Similarly, at

allocation step #3, if additional bits are allocated to the quantizer the combined audio signal and MNR value will lie proximate line 3208.

5       With reference to Fig. 32B, at allocation step #1 the combined audio and MNR level will lie proximate line 3210. At step #2, the it will be reduced to level 3211, and at allocation step #3, it will fall to line 3212. At allocation step 4, sufficient bits will be allocated to the quantizer to reduce the combined signal and MNR value to level 3213 which falls below the GMT min at point 3205.

10       The bit allocation routine as discussed above, progresses through the allocation steps until the combined signal and MNR value (hereafter the quantizing value) falls below the minimum of the GMT. During each innervation through the bit allocation routine, when the quantizing value is greater than the GMT min, the  
15       deallocation table is updated to include the difference value between the minimum of the GMT and the MNR value. Thus, the deallocation table of Fig. 32 stores the channel and subband for each difference value. In the present example, the deallocation table records for subband 3 (Fig. 32A) the difference value 3db  
20       which represents the distance between the minimum of the GMT at point 3204 and the quantization level at point 3207 above the GMT. The table also stores the allocation step associated with the quantization value at line 3207. The deallocation table also stores an element for subband 7 which represents the difference



value between the minimum of the GMT and the quantization level corresponding to line 3212.

During the deallocation routine, the deallocation table is resorted to place with the difference values in ascending order, such that the first element in the table corresponds to the subband with the least difference value between the minimum GMT and quantization level of the next closest MNR value. The quantizer corresponding to subband 7 is deallocated, such that the number of bits assign thereto is reduced from the number of bits corresponding to step #4 (line 3213) to the number of bits corresponding to step #3 (line 3212). Thus, the deallocation routine subtracts bits from the subband which will introduce the least amount of noise above the GMT for that subband. Once the subband 7 has been deallocated, the difference value is recalculated for the next preceding step (corresponding to MNR at line 3211). This new difference value is stored in the deallocation table along with its corresponding allocation step. If the number of bits deallocated during the first pass through this process is insufficient to lower the total allocated bits below the available bit pool maximum, than the processes repeated. In a second innervation, the quantizer corresponding to subband 3 would be reallocated with fewer bits corresponding to allocation step #2 (line 3207). This process is repeated until the total allocated bits falls within the available bit pool.

### Basic Components and CODEC System

Figure 1 illustrates a high level block diagram of a CODEC 1. Figure 1 shows an encoder digital signal processor (DSP) 1, a decoder DSP 2, an LED DSP 95, an asynchronous multiplexer 3, an asynchronous demultiplexer 6, at least one digital interface module (DIM) 7 connected to the encoder output, at least one DIM 8 connected to the decoder input, a loopback control module 9, and a control processor 5. The encoder 1 inputs digital signals and timing signals and outputs compressed audio bit streams. The decoder 2 similarly inputs compressed audio bit streams and timing signals and outputs decompressed digital signals.

The CODEC 1 is capable of holding several audio compression algorithms (e.g. ISO MPEG and G.722). These and other algorithms might be downloaded into the CODEC from ISDN and thus future upgrades are simple and effortless to install. This creates an extremely versatile CODEC that is resistant to obsolescence. This should be contrasted to the ROM type of upgrade procedure currently employed by most CODEC manufacturers.

The CODEC 1 may also use a unique compression technique which is explained below and is described in the attached Software Appendix. This compression technique also uses an increased number of psycho-acoustic parameters to facilitate even more efficient compression and decompression of audio bit streams. These additional parameters are described above.

The CODEC 1 also contains a control processor 5 for receiving and processing control commands. These commands are conveyed to the various CODEC 1 components by a line 51. These commands might be entered by a user via front panel key pads such as 15, 152, and 154, as shown in Figures 5, 6, and 7. Keypad commands enter processor 5 through a line 52. The keypad also allows the user to navigate through a menu tree of command choices which fall into the general categories of common commands, encoder commands, decoder commands, and maintenance commands. Such menu choices are displayed on a Front Panel LCD display (not shown) via signals from a processor 5 on a line 58. (See LCD Menu Summary of commands, Chap 8 of CODEC manual, attached to the end of this specification before the claims). The LCD display might also be used for characters to show responses to front panel user commands as well as spontaneous messages such as incoming call connect directives. Additionally, the LCD display may be used to display graphical information.

The CODEC processor 5 may receive commands from a front panel remote control panel (RS232 interface format) and enter the processor 5 through the line 54. A front panel remote control allows computer access to all internal functions of the CODEC 1. Front panel remote control is especially useful for applications that need quick access via a palm top or lap top computer. This frequently occurs in control rooms where there are many CODECs in equipment racks serving different functions. A full complement of

remote control commands exists to facilitate control of the CODEC 1 (See the listing of remote control commands from the "cdqPRIMA" operating manual, Chapter 9, attached to the end of specification).

5        Referring again to Figure 2, this more detailed block diagram of CODEC 1 shows external front panel remote control data interacting with Front Panel Remote Control UART 178 via a line 54. UART 178 is controlled by the Control Micro 5 via a control network line 155.

10        The CODEC 1 also provides a rear panel remote control port which uses either RS232 or RS485 interface formats. The RS485 port may be either a 2 or 4 wire interface. A rear panel remote control also allows computer access to all the internal functions of the CODEC 1. Rear panel remote control is especially useful for  
15        applications which need permanent access to the CODEC 1 via computer control. This frequently occurs when the CODEC 1 is remotely located from the control room. The electrical interface choice is controlled by a command entered through remote control or a keypad.

20        Referring again to Figure 2, this more detailed block diagram of the CODEC 1 shows external rear panel remote control data interacting with Remote Control UART 18 via line 56. UART 18 is controlled by Control Micro 5 via the control network line 155. The CODEC also includes a Front Panel LED display 3, examples of  
25        which are shown in Figures 11 and 12. This includes a set of

Status, Encoder, and Decoder LED's to show the status of various CODEC functions, for instance which compression algorithm is being used, and/or whether error conditions exist. The Status 31, Encoder 32, and Decoder 33 groups of LED's might be independently  
5 dimmed to allow emphasis of a particular group.

Referring again to Figure 1, signals from control processor 5 enter LED DSP 95 through the line 51. These control signals are processed by a LED DSP 95 and drive a LED display 3 (Figures 11 and 12) via a line 96.

10 A LED display 3 also shows peak and average level indications for the encoder 32 (left and right channels) and the decoder 34 (left and right channels). Each LED represents 2 dB of signal level and the maximum level is labeled dB. This maximum level is the highest level permissible at the input or at the output of the  
15 CODEC. All levels are measured relative to this maximum level. The level LED's display a 4 dB audio range. A peak hold feature of the level LED's shows the highest level of any audio sample. This value is instantly registered and the single peak level LED moves to the value representing this signal. If the peak level of all  
20 future signals are smaller, then the peak LED slowly decays to the new peak level. The peak level LED utilizes a fast attack and slow decay operation. The LED display 3 also includes a level display to show stereo image 36 which is used to display the position of the stereo image. This is useful when setting the levels of the  
25 left and right channels to insure the proper balance. Also

included is a correlation level display 38 which is used to check if the left and right channels are correlated. If the left and right channels are correlated, then they might be mixed to mono. The level LED's might also be used to display a scrolling message.

5 Referring again to Figure 2, this more detailed block diagram of CODEC 1 shows the LED DSP 95 driving a LED Array 125 via a connection 96. As also shown, the LED DSP 95 is controlled by the Control Micro 5 via the control network line 155. The DSP 95 also drives an Headphone (Hp) D/A Converter 98 via a connection 97. A  
10 converter 98 then outputs this analog signal via a connector 99 to external headphones (not shown). The headphones allow the user to monitor both the input and output signals of the CODEC 1. Figures 11 and 12 show headphone indicators 31 at the far right of the level displays to denote the signal output to the headphones. If  
15 both LED's are illuminated, then the left audio channel is output to the left earphone and the right audio channel is output to the right earphone. If only the left LED is illuminated, the left audio channel is output to both the left and right headphone. Similarly, if only the right LED is illuminated, the right audio  
20 channel is output to both the left and right headphone.

#### Analog Inputs and Outputs

Figure 2 shows a more detailed block diagram of the CODEC 1 structure. Referring to Figures 1 and 2, the left audio signal 12

and the right audio signal 14 are external analog inputs which are fed into an Analog to Digital (A/D) Converter 1, and converted into digital signals on a line 11. Similarly digital audio output signals on a line 121 are converted from Digital to Analog (D/A) via a converter 15. The converters 1 and 15 use an 18 bit format. The analog sections of the CODEC are set to +18 dBu maximum input levels. Other analog input and output levels might used.

#### Direct Digital Inputs and Outputs

Referring again to Figure 1, the CODEC 1 also allows for direct input of digital audio information via an AES/EBU digital audio interface on line 16 into encoder 1. The decoder 2 similarly outputs decoded, decompressed digital audio information on AES/EBU output line 22. These interfaces allow for interconnection of equipment without the need for A/D conversions. It is always desirable to reduce the number of A/D conversions since each time this conversion is performed, noise is generated. These interfaces might use a DB9 or XLR connectors.

AES/EBU digital input and output rates might vary and therefore such rates are converted, or adapted, by a Sample Rate Converter 11, to eliminate any digital clock problems. The A/D Converter 1 signals are similarly converted, or adapted, by a Sample Rate Converter 11 before entering the encoder 1. Because of the rate adapters, the input/output digital rates are not required to be the same as the internal rates. For example, it is possible

to input 44.1 kHz AES/EBU digital audio input and ask the CODEC 1 to perform compression at 48, 44.1 or 32 kHz (by using the front panel LCD display or a remote control command). This is possible because of the digital rate adapters. Similarly, digital audio input sources might be 32, 44.1, or 48 kHz. These input sampling rates are automatically sensed and rate adapted. The compression technique at the encoder determines the internal digital sampling rate at the decoder, and a control command is used to set this rate. The AES/EBU digital output sampling rate from the decoder is also set via a control command and might be a variety of values.

The digital audio is output from the decoder at the sampling rate specified in the header. This rate might then be converted to other rates via the Sample Rate Convertor 12. The Sample Rate Convertors 11, 12 are capable of sampling rate changes between .51 and 1.99. For example, if the receiver received a bit stream that indicated that the sampling rate was 24 kHz, then the output sampling rate could be set to 32 or 44 kHz but not 48 kHz since 48 kHz would be a sampling rate conversion of 2. to 1. This is out of the range of the sampling rate converter. The allowed output sampling rates include 29.5, 32, 44.1, and 48 kHz. Other direct digital I/O formats might include, for example, SPDIF or Optical.

The encoder 1 receives direct digital input via a connector on the rear panel (line 16). Analog or digital signals (but not both simultaneously) may be input into the CODEC 1 as selected by a front panel switch. If the digital input is selected, the CODEC 1



locks to the incoming AES/EBU input and displays the lock condition via a front panel LED. If digital audio input is selected, an AES phase-lock loop (PLL) is used to lock onto the signal. Accordingly, the AES PLL lock light must be illuminated before  
5 audio is accepted for encoding. In normal operation, the CODEC 1 locks its internal clocks to the clock of the telephone network. For loopback (discussed below), the CODEC 1 locks its clocks to an internal clock. In either case, the clock used by the CODEC 1 is not precisely the same frequency as the AES/EBU input. To prevent  
10 slips from occurring due to the presence of two master clocks, a rate synchronizer is built into the encoder section to perform the necessary rate conversion between the two clocks.

The decoder 2 outputs direct digital signals via a rear panel connector (line 22). Additionally, the decoder may be synchronized  
15 to an external clock by an additional connector (SYNC, line 24) on the rear panel. Referring also to Figure 8, a block diagram is shown of the decoder output timing with the AES/EBU SYNC (line 24) disabled or not present during normal timing. If no input is present on the decoder AES/EBU SYNC input line 24 (Figure 1), then  
20 the output AES/EBU digital audio is generated by the internal clock source 2 that is either at the telephone or internal clock rate. Figure 9 additionally shows a block diagram of the decoder output timing with the AES/EBU SYNC disabled or not present, and using internal crystal timing.

Referring to Figure 1, a block diagram is shown of the decoder output timing with the AES/EBU SYNC (line 24) enabled and present using AES timing. If the SYNC input is present, then the digital audio output is generated at the frequency of the SYNC input via the clock generator 25 being fed into the rate adaptor 252. This adapted rate is used by the D/A Converter 254, as well as the AES/EBU transmitter and receiver units 256, 258. The presence of a valid sync source is indicated by illumination of the front panel AES PLL LED. The sync frequency may be slightly different from that of the CODEC 1 clock source and again the rate synchronism is performed to prevent any undesired slips in the digital audio output. The SYNC input is assumed to be an AES/EBU signal with or without data present. The CODEC 1 only uses framing for frequency and sync determination.

Referring again to Figure 2, this more detailed block diagram of CODEC 1 shows external digital input 16 entering AES/EBU receiver 13. The receiver output 14 then enters the Sample Rate Converter 11 and the rate is converted, if necessary, as described above. The converter 11 then feeds the rate adjusted bit stream via a line 111 into the encoder 1 for coding and compression.

Conversely, Figure 2 also shows the Decoder DSP 2 outputting a decoded and decompressed bit stream via a line 123 into the Sample Rate Converter 12. The converter 12 adapts the rate, if necessary, as described above and outputs the rate adjusted bit stream via line 122 into a AES/EBU Transmitter 126. The

transmitter 126 then outputs the digital signal through an external connection 22.

Figure 2 also shows the AES/EBU digital synchronous input line 24 leading into a AES/EBU Receiver 146. The receiver 146 routes the received SYNC input data into the Sample Rate Converter 12 via a line 147. The converter 12 uses this SYNC input for rate adapting as described above.

#### Asynchronous Ancillary Data

The CODEC 1 is also capable of handling a variety of ancillary data in addition to primary audio data. The audio packet, for instance, consists of a header, audio data, and ancillary data. If the sampling rate is 48 KHz, then the length of each packet is 24 milliseconds. The header consists of a 12 bit framing pattern, followed by various bits which indicate, among other things, the data rate, sampling rate, and emphasis. These header bits are protected by an optional 16 bit CRC. The header is followed by audio data which describes the compressed audio signal. Any remaining bits in the packet are considered ancillary data.

Referring again to Figure 1, the CODEC 1 provides for transmission of ancillary data via an asynchronous, bi-directional RS-232 input interface 39, and an output interface 62. These interfaces provide a transparent channel for the transmission of 8 data bits. The data format is 1 start bit, 8 data bits, 1 stop bit and no parity bits. A maximum data rate might be selected by the

control processor 5. This interface is capable of transmitting at the maximum data rate selected for the encoder 1 and the decoder 2 and thus no data pacing such as XON/XOFF or CTS/RTS are provided.

5 The RS-232 data rates might be set from 3 to 19,2 bps. The use of the ancillary data channel decreases the number of bits available to the audio channel. The reduction of the audio bits only occurs if ancillary data is actually present. The data rate might be thought of as a maximum data rate and if there is no ancillary data present, then no ancillary data bits are  
10 transmitted. A typical example of this situation occurs when the CODEC 1 is connected to a terminal; when the user types a character the character is sent to the decoder at the bit rate specified.

The setting of the decoder baud rate selection dip switches is  
15 done by considering the setting of the encoder. The decoder baud rate must be an equal or higher baud rate relative to the encoder. For example, it is possible to set the decoder ancillary baud rate to 9,6 baud. In this case, the encoder baud rate may be set to any value from 3 to 9,6 but not 19,2. If the decoder baud rate is set  
20 to a higher rate than the encoder, the data will burst out at the decoder's baud rate. The maximum sustained baud rate is therefore controlled by the encoder.

The compression technique for the transmission of ancillary data is as follows: the encoder looks, during each 24 millisecond  
25 frame interval, to see if any ancillary data is in its input

buffer. If there are characters in the encoder's input buffer, then the maximum number of characters consistent with the selected baud rate are sent. During a 24 millisecond period, the table below shows the maximum number of characters per frame (at 48 kHz sampling rate) sent for each baud rate.

<u>BIT RATE</u>	<u>NUMBER OF CHARACTERS</u>
3	1
12	3
24	6
36	9
48	12
72	18
96	24
192	47

The CODEC 1 provides no error detection or correction for the ancillary data. The user assumes the responsibility for the error control strategy of this data. For example, at an error rate of  $1e-5$  (which is relatively high) and an ancillary data rate of 12 baud, 1 out of every 83 characters will be received in error. Standard computer data communication protocol techniques might be used to maintain data integrity. When designing an error protection strategy, it must be remembered that the CODEC 1 may occasionally repeat the last 24 milliseconds of audio under certain error conditions. The effect on audio is nearly imperceptible. However, the ancillary data is not repeated.

The format of the ancillary data is user defined. The present invention utilizes two formats for the ancillary data. The first format treats the entire data stream as one logical (and physical) stream of data. The second format allows for multiplexing of

various logical and diverse data streams into one physical data stream. For example, switch closure, RS232, and time code data are all multiplexed into a single physical data stream and placed in the ancillary data stream of the ISO MPEG packet.

5        Figure 1 shows a high level diagram of the asynchronous multiplexer (MUX) 3 in relation to the other CODEC components. Figure 3 shows an isolated diagram of the multiplexer 3 in relation to encoder 1. The data rate for the multiplexer is set by software command (via remote control connections or keypad entry). A  
10       software command also controls a switch 34 (Figure 1) which routes the ancillary data through multiplexer 3. Multiplexer output line 36 routes the multiplexed data into the encoder input line 38. Alternatively, if the switch 34 is in the other position, ancillary data will be routed directly to the encoder input line 38 via the  
15       input line 32 without multiplexing. When the multiplexer 3 is used, Figure 1 shows signals from input sources such as RS485 (line 31), RS232 (line 33), contact closures (line 35), time codes (line 37), and ancillary data -- RS232 (line 39). Figure 3 shows similar inputs into multiplexer 3. These ancillary inputs are used as  
20       follows:

      The RS232 I/O connector is used to provide an additional port into the data multiplexer. It might be thought of as a second RS232 ancillary port. The RS485 I/O connector is used to provide an additional type of port into the data multiplexer. It is a  
25       dedicated RS485 port and might be used to control RS485 equipment.

Contact closure inputs 3 allow simple ON/OFF switches to be interfaced into the CODEC 1. The contact closure inputs 3 are electrically isolated from the internal circuitry by optical isolators. A plurality of optical isolated I/O lines and/or  
5 contact closure lines might be used. Additionally, the time code inputs allow transmission of timecode at rates of 24, 25, 29, and 3 frames per second.

Referring again to Figure 3, the Ancillary Data Multiplexer 3 multiplexes the various inputs into a composite ancillary data stream for routing to encoder input line 38. The encoder 1 then  
10 processes the digital audio signals (e.g. converted left and right analog inputs, AES/EBU, SPDIF, or optical) and the ancillary data stream (e.g. multiplexed composite or direct) into a compressed audio bit stream. In Figure 3, an ISO/MPEG encoder 1 is shown,  
15 with the digital audio left and right signals, as well as a composite ancillary data stream, being processed by the ISO/MPEG encoder 1 into a resulting ISO/MPEG audio bit stream. Other compression techniques besides ISO/MPEG could similarly be illustrated.

20 Conversely, a block diagram is shown in Figure 4 wherein the ISO/MPEG Audio Bit Stream enters an ISO MPEG Decoder 2 on line 22. The bit stream is decoded (decompressed) and the ancillary data is separated from the audio data. The composite ancillary data stream enters the Ancillary Data De-Multiplexer 6 through line 23. The  
25 Ancillary data is de-multiplexed into its component parts of

Ancillary, RS232, RS485, Time Code, and Relay Contact data, as shown by lines 61, 63, 65, 67, and 69. The audio data (left and right) is output on lines 26 and 28. A software command also controls a switch 64 (Figure 1) that might route the ancillary data out of decoder 2, through the de-multiplexer 6, through line 66, and out to ancillary data line 69. Alternatively, the ancillary data might be routed directly from decoder output line 23, through line 62, and out line 69 -- without multiplexing.

Referring again to Figure 2, this more detailed block diagram of CODEC 1 shows external ancillary data entering the ancillary data switch 16 via line 39 and exiting switch 16 via line 69. (See lines 39, 69 and switches 34, 64 in Figure 1). Switch 16 interacts with Ancillary Data UART (Universal Asynchronous Receiver Transmitter) via connections 164 and 165. Switch 16 also interacts with DSP Ancillary Data UART 169 via connections 166 and 167. The resulting data is sent through Switch 16 to encoder 1 via connection 162. Decoded ancillary data is sent through Switch 16 from decoder 2 via connection 163. Switch 16, Ancillary Data UART 168, and DSP Ancillary Data UART are controlled by Control Micro 5 via control network line 155.

Figure 2 also details the following ancillary data connections: External RS232 data is shown entering RS232 UART 17 via line 33 and exiting UART 17 via line 69. External Time Code Data is shown entering SMPTE Time Code Interface 172 via line 37 and exiting via line 67. Time Code Data is subsequently shown



interacting with Time Code UART 174 via lines 173, 175. External RS485 data is shown entering RS485 UART 176 via line 31 and exiting via line 61. External Optical inputs are shown entering Control micro network 155 via line 35. Relay outputs are shown exiting Control micro network 155 via line 65. UARTS 17, 174, 176, and Time Code Interface 172 are controlled by Control Micro 5 via control network line 155.

Ancillary data can prove to be extremely valuable because it allows the CODEC user to transmit control and message information to and from RS232 and RS485 equipment, on either end of the transmission channel, via the same compressed digital bit stream as used by the audio signal component. The user might also send time code information and facilitate the control of relay contacts. More importantly, the use of ancillary data does not adversely affect the ability to transmit a sufficiently large amount of primary audio data.

#### **Synchronous Ancillary Data**

Referring again to Figure 1, the CODEC 1 also provides a synchronous ancillary input data line 18 and output data line 25. The synchronous connections might exist separately (as shown in Figures 1 and 2) or as part of a multi-functional input line (e.g. optical isolated I/O, relay I/O and synchronous ancillary data I/O share a common line -- not shown). This data port is an RS232 interface, and might also include RS422 and/or RS485 capabilities.

### Digital Interface Modules and Loopback Control

Referring again to Figure 1, encoder 2 outputs a compressed audio bit stream through line 4 (and possibly more lines) into at least one DIM 7. These modules might include, for example, the types X.21/RS422, V.35, and/or TA. These modules output the digital signals for use and/or transmission by equipment external to the CODEC. Similarly, DIM 8 is connected to decoder 2 through line 81. DIM 8, using similar type modules as DIM 7, collects the external digital signals for transmission to decoder 2.

Referring again to Figure 2, this more detailed block diagram of CODEC 1 shows the compressed bit stream entering H.221 DSP 19 via line 191. DSP processes the bit stream and transfers the data, via line 192, to at least one DIM (Module types shown as 198). DIM 198 interacts with TA Control UART 193 via lines 194, 195, and with Decoder DSP 2 via line 197. DIM 192 then outputs external data via line 71 and inputs external data via line 81. As discussed above, this external data is then used by external equipment such as transmitters and receivers.

Before any connection is made to the outside world, the DIMs in CODEC 1 must be defined. If the DIMs are rearranged, then the CODEC must be notified via remote control software command (through the keypad or remote control interface). For DIMs that dial outside networks, two methods of dialing exist. They are single line dialing and multiple line dialing (speed dialing). For either mode of dialing it is possible to enable automatic reconnect. This



feature allows the automatic reconnection of a dropped line. If auto reconnect is enabled when a line is dialed, then it will be reconnected if either the far end disconnected the call, or the network drops the call. If the calling end drops the call, the line will not be automatically reconnected. This feature also allows the DIM to automatically dial an ISDN network if, for instance, a satellite connection is lost.

The CODEC 1 provides for two types of loopback through loopback control module 9. Loopback is an important feature for CODEC testing purposes. The first type is a system loopback and the second is a digital interface loopback. The system loopback is an internal loopback which loops back all the digital interfaces and is set by one software command. The second type of loopback allows the user to select individual digital interface modules for loopback. Loopback control might also be used to cause the internal CODEC clock to supply the digital data clocks.

#### Satellite Receiver Interfaced with CODEC

Referring to Figure 13, another embodiment of the disclosed invention allows for the transmission of other information besides audio, including, video, text, and graphics. In this embodiment, the digital line inputs 41 are preferably replaced with a satellite antenna 46. The digital interface module 42 (or satellite receiver module) receives digital signals that are transmitted to it by the satellite antenna 46. The digital signals, which are streams of

data bits, are then transferred to a decoder 42. The decoder decompresses the bits, whether they are audio, video, text, or graphic, and directs them to the appropriate output.

Preferably, the digital interface module 42 has the ability to store digital information. In this alternate embodiment, the digital interface module (satellite receiver module) is preferably a receiver called a "daX". Such a receiver is available commercially under the name "daX" from Virtual Express Communications in Reno, Nevada. In this embodiment, the decoder preferably would have the capability to decompress or decode other types of compressed information such as video, text, and graphics. This could be facilitated by downloading the required compression techniques into the CODEC 1 as described above.

In its operation, the satellite antenna 46 might receive digital information from various sources including a remote CODEC or a remote daX (not shown), and transfer the information to the daX receiver 42. The daX DIM 44 might also act as a switching mechanism to route the digital bit streams to different places. It might direct information received from the satellite directly to the decoder, via line 4, for decompression and immediate output. The received data from the satellite receiver 42 might alternatively be directed through the daX DIM 44 to the daX 45 via line 43 for storage and later retrieval. The digital interface module 44 might then direct these stored data bits from the daX 45 to the decoder 42 via path 4 for decoding and subsequent output.

This embodiment also preferably allows for simultaneous storage of digital information in the DAX via path 43 and for immediate decoding of digital information via line 4 through the decoder 42.

5 While few preferred embodiments of the invention have been described hereinabove, those of ordinary skill in the art will recognize that these embodiments may be modified and altered without departing from the central spirit and scope of the invention. Thus, the embodiments described hereinabove are to be considered in all respects as illustrative and not restrictive, the  
10 scope of the invention being indicated by the appended claims, rather than by the foregoing descriptions, and all changes which come within the meaning and range of equivalency of the claims are intended to be embraced herein.

15 **CODEC SOFTWARE COMMAND DESCRIPTIONS  
AND CODEC OPERATIONS MANUAL**

Attached hereto are relevant portions of the operation manual for the CODEC 1, which includes detailed descriptions of the remote control software commands and their usage with the CODEC 1 described above.

```

opt    ic
;
; (c) 1994. Copyright Corporate Computer Systems, Inc. All rights
; reserved.
; \UXCODE\bitalloc.asm:
;
; This routine is used to allocate the bits.
; It allocates at least some bits to all sub-bands with a positive
; SMR.
; It allocates in three phases:
;   A. allocate all sub-bands until they are all below
;       the Global Masking Threshold (regardless as to how many
;       bits it takes)
;       note 1. a limit (sub-band boundary) is set which requires
;               all sub-bands up to the boundary require at least
;               index 1 be allocated even if the signal is already
;               below the Global Masking Threshold. (This provides
;               a noticeable improvement in continuity of sound)
;       note 2. For JOINT stereo framing,
;               a. a 1st pass is made thru Phase A with the frame type
;
;                   set to FULL stereo to see if the framing bit pool
;                   can handle FULL stereo.
;                   The 1st sub-band (channel) that exceeds the bit pool
; as                   it is allocated for below the Global Masking
; Threshold           causes the 1st pass thru Phase A to be aborted
;                   indicating the a JOINT frame is necessary.
;                   b. JOINT framing uses the aborted sub-band to set the
;                       intensity sub-band boundary to 4, 8, 12 or 16.
;                       A new bit pool is determined based on this boundary.
;                       A call to the routines to calculate the JOINT Stereo
;                       arrays is made.
;                       A 2nd pass thru Phase A is made for a JOINT stereo
;                       frame and a new bit pool size.
; After Phase A is completed, a test is made to see if the bit
; pool
;   was overflowed by the allocation.
;   a. if the frame fits, Phase B is skipped and Phase C is done
;   b. otherwise, Phase B is required to selectively de-allocate
; the
;       best sub-band candidates.
;
; on entry
;   y: < stereo = flags:
;       (set on entry) bit 0 means stereo vs mono framing
;                       0 = stereo framing
;                       1 = mono framing
;       bit 1 is used to indicate left vs right channel
;                       0 = looping through left channel
; arrays
;       1 = looping through right channel arrays

```

```

; bit 2 is to simply indicate that joint stereo applies
; 0 = NOT joint stereo framing type
; 1 = IS joint stereo framing type
; bit 3 is to indicate the full stereo initial
allocation
; if joint stereo applies
; 0 = normal joint stereo allocation
; 1 = FULL STEREO initial joint stereo
allocation
; bit 4 is to simply indicate the stereo intensity
sub-band
; boundary has been reached if joint stereo applies
; 0 = NO sub-bands still below
intensity boundary
; 1 = sub-bands above intensity
boundary
; bit 5 is used as the FirstTime switch in an
allocation
; 0 = cleared if any allocations were
made
; 1 = no allocations made to any
sub-bands
; bit 6 is used for critical de-allocate and allocate
passes:
; with below masking threshold being a criteria
de-allocate:
; 0 = select from any sub-band channel
; 1 = select from only those below mask
allocate:
; 0 = there are sub-band channels not
below mask
; 1 = all sub-bands are below mask
; bit 7 is used for critical de-allocate and allocate
passes:
; de-allocate:
; 0 = select from any sub-band channel
; 1 = select from those with 2 or more allocation
allocate:
; 0 = are sub-bands not below hearing
thresh
; 1 = all sub-bands are below hearing
thresh
; bit 8 is used for critical de-allocate and allocate
passes:
; de-allocate:
; 0 = select from any sub-band channel
; 1 = select from any sub-band channel
allocate: for final pass after bit allocation timer
;
; 0 = timer interrupt not yet sensed
;
; 1 = timer interrupt was sensed
;
; bit 9 is to simply indicate that the sub-band limit

```

```

for
    allocating at least ONE position has been reached
    within a current loop:
        0 = NOT at sub-band limit
        1 = reached the sub-band limit
    bit 10 is to simply indicate that the maximum
sub-band for
    consideration for allocation has been reached
    within a current loop:
        0 = NOT at maximum sub-band limit
        1 = reached the maximum sub-band
limit
;
;   y:audbits = number of bits available for sbits, scale factors
and data
;   y:<usedsb = number of sub-bands actually used
;   y:<maxsubs = MAXSUBBANDS at sampling rate and bit rate
;   y:sibound = stereo intensity sub-band boundary
;   y:stintns = stereo intensity sub-band boundary code for frame
header
;   y:limitsb = number of sub-bands requiring at least one
allocation
;   y:bandcnt = decremented sub-band counter intensity boundary
check
;   y:frmtype = framing type specified by external dip switches
;   y:opfrtyp = current frame output type (Joint may upgrade frame
to full)
;   y:<qtalloc = timer interrupt set to signal quit allocation
loops
;   r0 = addr of the SBits array left and right channel (x memory)
;   r1 = addr of MinMasking Db array left and right channel (x
memory)
;   r2 = addr of SubBandMax array left and right channel (x
memory)
;   r4 = addr of the SubBandPosition array left and right channel
(x memory)
;   r5 = addr of the SubBandIndex array left and right channel (x
memory)
;
; on exit
;   a = destroyed
;   b = destroyed
;   x0 = destroyed
;   x1 = destroyed
;   y0 = destroyed
;   y1 = destroyed
;   r3 = destroyed
;   r6 = destroyed
;   n0 = destroyed
;   n1 = destroyed
;   n2 = destroyed
;   n3 = destroyed
;   n4 = destroyed
;   n5 = destroyed

```





```

; n6 = destroyed
;
; AtLimit array by sub-bands (left for 32, then right for 32):
;   bit 0 set when allocation is below the masking threshold
;   bit 1 set when allocation is below the threshold of
hearing
;   bit 2 set when allocation is at the limit of maximum
position
;   or there are not enough bits to allocate
;   the sub-band further
;
include 'def.asm'
include 'box_ctl.asm'
;
section lowmisc
;
xdef BitsAdd
xdef BPosAdd
xdef BInxAdd
xdef AllwAdd
xdef MNRsub
xdef MNRsb
xdef MNRchan
xdef MNRmin
xdef MNRinx
xdef MNRpos
xdef AvlBits
xdef TotBits
xdef HldBits
xdef count
xdef svereg
xdef jntflag
;
org yli:
stbitalloc_yli
;
BitsAdd    ds    1    ;save address of SBits array
BPosAdd    ds    1    ;save address of SBPosition array
BInxAdd    ds    1    ;save address of SBIndex array
AllwAdd    ds    1    ;save addr of applicable Allowed
table
MNRsub     ds    1    ;count of entries in de-allocate
tables
MNRsb      ds    1    ;curr sub-band for allocation
MNRchan    ds    1    ;channel of curr sub-band for
allocation
MNRmin     ds    1    ;value of curr sub-band for
allocation
MNRinx     ds    1    ;new index for selected sub-band
MNRpos     ds    1    ;new allowed position for selected sb
AvlBits    ds    1    ;available bits to allocate
TotBits    ds    1    ;current bit count allocated
HldBits    ds    1    ;sub-band critical allocation
count      ds    1    ;sub-band counter

```

```
svereg    ds    1    ;save register for restoring
jntflag    ds    1    ;bits to control joint functions:
; 0 - for demand bits pass
; 0 - not at available bits yet
; 1 - reached end of avail bits
```

```
endbitalloc_yli
endsec
```

```
section highmisc
```

```
    xdef    UsedSBs
    xdef bitallocR7Save
    xdef bitallocN7Save
    xdef bitallocM7Save
```

```
    org     xhe:
stbitalloc_xhe
```

```
;This array is the counters for sub-bands with assigned indices
;If a sub-band starts out below the Global Masking Threshold it
takes
;a certain number of consecutive frames before it is skipped. Until
that
;count down (SUBBANDSCTDOWN) reaches zero, the sub-band will
receive at
;least one allocation.
```

```
UsedSBs    ds    NUMSUBBANDS*2
```

```
;these save variables for exclusive use by bitalloc only
```

```
bitallocR7Save ds    1
bitallocN7Save ds    1
bitallocM7Save ds    1
```

```
endbitalloc_xhe
endsec
```

```
section highmisc
xdef strt sin
xdef endsin
xdef uselmsb
xdef demand
xdef jntadj
xdef jntsub
xdef boundlst
xdef isocdelst
xdef jntfrms
xdef jfrmcnt
xdef UsedSBReg
xdef MaxPos
xdef ndatabit
    xdef    NDataBit
```

```

xdef NSKFBits
xdef SNR

org yhe:
stbitalloc_yhe

;sub-band range for a possible sine wave in the current channel
;if not a sine wave, these values are -1

strtsin ds 1 ;start sub-band span for sine wave
endsin ds 1 ;end sub-band span for sine wave
uselmsb ds 1 ;use LIMITSUBBANDS not greater thane
y:<usedsb
demand ds 1 ;demand bits
jntadj ds 1
jntsub ds 1
boundlst ds 1 ;boundary to use for jntfrms
isocdelst ds 1 ;intensity boundary ISO code for boundlst
jntfrms ds 1 ;count of frames to maintain
jfrmcnt ds 1 ;frame counter
UsedSBReg ds 1 ;current addr into UsedSbs counters array
MaxPos ds 1 ;Max Position per selected Allowed table

;This is the addr of the selected table, ISO or COMPRESS,
; for the number of bits for data allocation by position

ndatabit ds 1 ;addr of ISO or COMPRESS NDataBit tbl

;This is the ISO table for the number of bits for data allocation
by position

NDataBit
dc 0*NUMPERSUBBAND ;index = 0, no transmit =
0 bits
dc 5*NUMPERSUBBAND ;index = 1, packed =
60 bits
dc 7*NUMPERSUBBAND ;index = 2, packed =
84 bits
dc 9*NUMPERSUBBAND ;index = 3 =
108 bits
dc 10*NUMPERSUBBAND ;index = 4, packed =
120 bits
dc 12*NUMPERSUBBAND ;index = 5 =
144 bits
dc 15*NUMPERSUBBAND ;index = 6 =
180 bits
dc 18*NUMPERSUBBAND ;index = 7 =
216 bits
dc 21*NUMPERSUBBAND ;index = 8 =
252 bits
dc 24*NUMPERSUBBAND ;index = 9 =
288 bits
dc 27*NUMPERSUBBAND ;index = 10 =
324 bits

```

	dc	30*NUMPERSUBBAND	;index = 11	=
360 bits	dc	33*NUMPERSUBBAND	;index = 12	=
396 bits	dc	36*NUMPERSUBBAND	;index = 13	=
432 bits	dc	39*NUMPERSUBBAND	;index = 14	=
468 bits	dc	42*NUMPERSUBBAND	;index = 15	=
504 bits	dc	45*NUMPERSUBBAND	;index = 16	=
540 bits	dc	48*NUMPERSUBBAND	;index = 17	=
576 bits				

;This is the COMPRESS table for number of bits for data allocation by position

	dc	0*NUMPERSUBBAND	;index = 0, no transmit	=
0 bits	dc	4*NUMPERSUBBAND	;index = 1, packed	=
48 bits	dc	6*NUMPERSUBBAND	;index = 2, packed	=
72 bits	dc	8*NUMPERSUBBAND	;index = 3, packed	=
96 bits	dc	10*NUMPERSUBBAND	;index = 4, packed	=
120 bits	dc	12*NUMPERSUBBAND	;index = 5	=
144 bits	dc	15*NUMPERSUBBAND	;index = 6	=
180 bits	dc	18*NUMPERSUBBAND	;index = 7	=
216 bits	dc	21*NUMPERSUBBAND	;index = 8	=
252 bits	dc	24*NUMPERSUBBAND	;index = 9	=
288 bits	dc	27*NUMPERSUBBAND	;index = 10	=
324 bits	dc	30*NUMPERSUBBAND	;index = 11	=
360 bits	dc	33*NUMPERSUBBAND	;index = 12	=
396 bits	dc	36*NUMPERSUBBAND	;index = 13	=
432 bits	dc	39*NUMPERSUBBAND	;index = 14	=
468 bits	dc	42*NUMPERSUBBAND	;index = 15	=
504 bits	dc	45*NUMPERSUBBAND	;index = 16	=
540 bits	dc	48*NUMPERSUBBAND	;index = 17	=
576 bits				

86

BAD ORIGINAL

```

;Each sub-band, if it is transmitted, must send scale factors. The
;Sbit patterns determine how many different scale factors are
transmitted.
;The number of scale factors transmitted may be 0, 1, 2 or 3. Each
scale
;factor requires 6 bits.

```

```

;Sbit patterns
; 00 Transmit all three scale factors 18 (3 * 6
bits)
; 01 Transmit the second two scale factors 12 (2 * 6
bits)
; 10 Transmit only one scale factor 6 (1 * 6
bits)
; 11 Transmit the first two scale factors 12 (2 * 6
bits)

```

```

;The NBits array is used to determine the number of bits to
allocate for the
;scale factors. NSBITS (the 2 bits for SBits code) are added to
account for
;all required scale factor bits (18+2,12+2,6+2,12+2).

```

```

NSKFBits
dc 20,14,8,14

```

```

;This is the table for Signal to Noise ratio by position

```

```

include '...\xlcode\snr.asm'

```

```

endbitalloc_yhe
endsec

```

```

org phe:

```

```

bitalloc

```

```

; bset WATCH_DOG ;tickle the dog
; OFF_BITALLOC_LED_CD ;tickle the led

```

```

;save register 7 and its attendants

```

```

move r7,x:bitallocR7Save
move n7,x:bitallocN7Save
move m7,x:bitallocM7Save

```

```

move #-1,m7 ;set to a linear buffer control

```

```

;Save the left and right channel array starting addresses

```

```

array move r0,y:BitsAdd ;save register of SBits
move r4,y:BPosAdd ;save register of

```

```

SubBandPosition array
    move    r5,y:BinxAdd      ;save register of
SubBandIndex array

;select the ISO or COMPRESS table for NDataBit:

    move #NDataBit,r5        ;standard ISO table
    move #19,n5              ;offset to COMPRESS table
    jclr #USE_COMPRESS,y:<cmprctl,_bita_05_A
    move (r5)+n5             ;select the COMPRESS table

_bita_05_A
    move r5,y:ndatabit      ;set addr of NDataBit table for alloc

;set up the MNR arrays for the left and right channels and the
joint channel
; if applicable

    move #SBMSr,r5          ;addr of Mask-to-Signal by sub-band
    move #NUMSUBBANDS,n5    ;offset to right channel values
    move r5,r3              ;addr of left chan Mask-to-Sig array
    move (r5)+n5            ;add offset to right channel
    move r5,r4              ;addr of right chan Mask-to-Sig array
    move (r5)+n5            ;add 2nd offset to joint channel
    move n5,n1              ;access right channel MinMsk values
    move n5,n2              ;access right channel SBMax values

;apply the safety factor

    move y:o_psych,y0       ;get the safety factor

;loop through the required sub-bands

    do    y:<usedsb,_bita_30_A
    move x:(r2+n2),x0        ;get right channel SBMax
    move x:(r1+n1),b         ;get right channel MinMsk
    sub x0,b x:(r2)+,x0      ;MinMask - SBMax = Mask-to-Signal
ratio
    sub y0,b x:(r1)+,a      ; & get left channel SBMax, incr nxt sb
    sub y0,b x:(r1)+,a      ; apply safety factor to right channel
    sub y0,b x:(r1)+,a      ; & get left channel MinMsk, incr nxt sb
    move b,x:(r4)+          ;store for test if below mask already
    sub x0,a                ;MinMask - SBMax = Mask-to-Signal ratio
    sub y0,a                ;apply safety factor to left channel
    move a,x:(r3)+          ;store for test if below mask already

;if doing joint stereo, develop the Joint Mask-to-Signal from the
lesser
; of the left and right channels

    jclr #JOINT_FRAMING,y:<stereo,_bita_20_A
    cmp a,b                 ;compare left and right MNR values
    jlt <_bita_10_A         ;b (right chan) is less, store that
one

```

```

        move a,x:(r5)+      ;otherwise store a (left chan) as less
        jmp  <_bita_20_A

_bita_10_A
        move b,x:(r5)+      ;b (right chan) is less, store that one

_bita_20_A
        nop

_bita_30_A                      ;END of y:<usedsb do loop

;set the working value for bits available for allocation
; NOTE: this value may be changed for JOINT stereo if the FULL
stereo
;       bit allocation for the frame CANNOT be handled
;       (for JOINT stereo,
;       y:audbits is the available bit count for FULL stereo)

        move y:audbits,x0      ;get standard available bit cnt
        move x0,y:AvlBits      ;store as working bit cnt

;save original array of used sub-band count down counters

        move #UsedSBs,r0
        move #SvUsedSBs,r1
        do  #NUMSUBBANDS*2,_bita_31_A
        move x:(r0)+,x0
        move x0,x:(r1)+

_bita_31_A

;initialize the bit allocation control flags in y:<stereo

        bclr #JOINT_at_FULL,y:<stereo ;init flag NOT at FULL

;if doing joint stereo,
; set flag for initial allocation to drive subbands to masking
; threshold to see if frame can handle full stereo

        jclr #JOINT_FRAMING,y:<stereo,_bita_40_A ;not joint frames,
continue
        bset #JOINT_at_FULL,y:<stereo ;set for initial Joint pass
        move #0,x1                  ;clear joint flag
        move x1,y:<jntflag           ;for joint demand bit rate ctl

_bita_40_A

;set usable LIMITSUBBANDS: if greater than y:<usedsb, use y:<usedsb

        move y:limitsb,x1          ;get static LIMITSUBBANDS
        move y:<usedsb,a           ;get the used sub-band cnt
        cmp  x1,a,x1,y:uselmsb     ;test limitsb vs usedsb
; & in case, set usable limistb
        jge  <_bita_41_A           ;if used > limit, continue

```

;used sub-band count is less the LIMITSUBBANDS, set to used sub-bands

move a,y:useImsb

\_bita\_41\_A

;(c) TotBits = 0; /\* start the bit allocation counter \*/

clr a #>1,x1 ;total bit used, x1 = 1 for start index

move a,y1 ;y1 = 0 to initialize

move a,y:TotBits

move a,y:count ;start the sub-band counter

bclr #AT\_LIMIT\_SUBBAND,y:<stereo ;NOT yet at sub-band limit ; which require at least 1 allocation

bclr #AT\_USED\_SUBBAND,y:<stereo ;NOT yet at sub-band maximum ; limit for coding used sub-bands

;initial allocation for all sub-bands;

1. that are within the use (less than UsedSubBands  
2. with a MinimumMasking to MaximumSignal above the masking threshold

move #SBMNRmax,r0 ;addr of de-alloc Max signal-noise

move #SBMSr,r1 ;addr of Mask-to-Signal by sub-band

move y:BitsAdd,r2 ;set register of SBits array

move y:AllwAdd,n3 ;init the current Allowed table

move y:BPosAdd,r4 ;set register of SubBandPosition

array move y:BinxAdd,r5 ;set register of SubBandIndex

array move #UsedSBs,r6 ;set start addr of used sub-band cnts

move r6,y:UsedSBReg ;set current (0) used sub-band cnt

addr move #AtLimit,r6 ;point to SubBandAtLimit array

;in case of joint stereo, clear the reached intensity sub-band boundary flag

move y:sibound,x0 ;joint stereo intensity sub-band

move x0,y:bandcnt ;bound subband decremented cnt

bclr #JOINT\_at\_SB\_BOUND,y:<stereo ;clear reached boundary sub-band

;initial allocation pass

;do all required sub-bands alternating between the left and right channels

; for the joint stereo 2nd pass make address alterations for joint arrays



```

do    #NUMSUBBANDS,_bita_230_A

;clear the n registers for the left channel reference

    clr  a    #0,n0          ;clear reg a to zero
                                ; & set n0 for left channel SBMNRmax
    move a,n1                ;SBMSr array
    move a,n2                ;SBits or Joint SBits array
    move a,n4                ;SBPos array
    move a,n5                ;SBIndx array
    move a,n6                ;AtLimit array
    bclr #LEFT_vs_RIGHT,y:<stereo    ;flag for left channel in
progress

;initialize for the possible presence of a sine wave in left
channel
;get the left xpsycho sine wave sub-bands to handle possible sine
wave

    move y:strtsinlft,a      ;start entry equals 1st sub-band
    move a,y:strtsin         ;isolate the starting sub-band
    move y:endsinlft,a       ;end entry equals last sub-band
    move a,y:endsin          ;isolate the ending sub-band

; if joint stereo does NOT apply, continue

    jclr #JOINT_FRAMING,y:<stereo,_bita_60_A

; if joint stereo upgraded to full, continue

    jset #JOINT_at_FULL,y:<stereo,_bita_60_A

; if doing joint stereo and have already switched over to joint
SBits array,
; but now have to adjust to 3rd set of SBMSr values

    jset #JOINT_at_SB_BOUND,y:<stereo,_bita_50_A

; see if the joint stereo intensity sub-boundary has been reached
; if not, continue at full stereo for these early sub-bands
; otherwise, switch over to the JointSBits

    move y:bandcnt,r3        ;get decrement sub-band ctr
    jsr  chkjoint            ;see if reached boundary
    move r3,y:bandcnt        ;save new decremented ctr
    jclr #JOINT_at_SB_BOUND,y:<stereo,_bita_60_A

    move #JntSBits,r2        ;shift over to Joint SBits array
    move y:count,n2          ;to offset to current sub-band
    nop
    move (r2)+n2              ;adj addr to current sub-band
    move #0,n2                ;reset to left channel

_bita_50_A

```

```

;we're at intensity sub-band limit
; shift over to Joint channel in SBMSr array
; 3rd set of sub-band values in n1

    move #NUMSUBBANDS*2,n1          ;Joint SBMSr values by sub-band

_bita_60_A

;process the current channel

    do #NUMCHANNELS,_bita_220_A

;initialize the pertinent sub-band values to 0

    move y1,x:(r6+n6)          ;clear allocated limit flag (AtLimit)
    move y1,x:(r5+n5)          ;clear allocated index (SBIndx)
    move y1,x:(r4+n4)          ;clear allocated position (SBPos)

;if we reached the used sub-band limit,
; take this one out of the picture completely

    jset #AT_USED_SUBBAND,y:<stereo,_bita_185_A

;if doing mono and we are processing the right channel,
; take this one out of the picture completely

    jclr #STEREO_vs_MONO,y:<stereo,_bita_70_A ;if doing stereo,
continue
    jset #LEFT_vs_RIGHT,y:<stereo,_bita_185_A ;if right, bag this
one

_bita_70_A
    jclr #JOINT_at_SB_BOUND,y:<stereo,_bita_80_A
    jset #LEFT_vs_RIGHT,y:<stereo,_bita_185_A ;right chan at
intensity
; take sub-band out of picture totally

_bita_80_A
    move y:count,y0          ;get current sub-band (00-31)

;see if we reached the used sub-band limit

    jset #LEFT_vs_RIGHT,y:<stereo,_bita_85_A ;left channel did
this
    move y:<usedsb,b          ;get count of used subbands for
testing
    cmp y0,b          ;see if sub-band not to be coded
    jgt <_bita_85_A          ;if not, continue
    bset #AT_USED_SUBBAND,y:<stereo ;just reached sub-band
maximum
    jmp <_bita_185_A          ;take completely out of use

_bita_85_A

```

```

;save current sub-band AtLimit addr for re-use after UsedSBs
counter processed
; and set address of current sub-band in use count down counter

```

```

move r6,y:svereg
move y:UsedSBReg,r6

```

```

;if we reached the sub-band limit for those requiring at least one
sub-band,
; see if we have anything to allocate to get below the Global
Masking Threshold

```

```

jset #AT_LIMIT_SUBBAND,y:<stereo,_bita_90_A

```

```

;see if at least one allocation is required regardless of signal to
noise ratio

```

```

jset #LEFT_vs_RIGHT,y:<stereo,_bita_95_A ;left channel did
this
move y:uselmsb,a ;get sub-band limit for at least 1
alloc
cmp y0,a ;if there is initial allocation
jgt <_bita_95_A ;continue
bset #AT_LIMIT_SUBBAND,y:<stereo ;just reached that limit
_bita_90_A

```

```

;if this channel has a sine wave, continue the allocation algorithm

```

```

move y:strtsin,a ;get start sub-band if sine wave
tst a ;if -1, no sine wave
jge <_bita_95_A ;if NOT -1 it's sine wave, continue

```

```

;otherwise, see if below Mask-to-Signal

```

```

move x:(r1+n1),a ;get sub-band's Mask-to-Signal ratio
tst a x:(r6+n6),a ;test Mast-to-Sig for positive value
; & get current count down value
jle <_bita_95_A ;if above masking thresh, init
counter

```

```

;test the used sub-band count down counter to see if this sub-band
; can be skipped from at least 1 allocation

```

```

tst a y:svereg,r6 ;see if zero and can be skipped
; & in case it can, reset AtLimit addr
jle <_bita_190_A ;counter = zero, set Below Mask flag

```

```

;decrement the count down counter and make 1 allocation

```

```

sub x1,a y:UsedSBReg,r6 ;decrement
; & set addr used sub-band counter
jmp <_bita_96_A ;update count down counter

```

\_bita\_95\_A

; initialize the used sub-band count down counter

```

;   move #>SUBBANDCNTDOWN,a
;   move y:z1_psych,a           ; get the count down value

```

\_bita\_96\_A

; update the used sub-band count down counter and reset AtLimit address reg

```

;   move a,x:(r6+n6)
;   move y:svereg,r6

```

; look for a sine wave in this channel

; and if so,  
; see if the current sub-band is within the sine wave sub-band range

; and if so,  
; force the allocation to the maximum

```

;   move y:strtsin,a           ; get start sub-band if sine wave
;   tst a y:count,y0          ; if -1, no sine wave
;                               ; & get current sub-band to test
;   jlt <_bita_97_A           ; if -1 not sine wave, continue
;   cmp y0,a y:endsin,a       ; current sub-band vs start sub-band
;                               ; & get end sub-band
;   jgt <_bita_180_A           ; if not yet reached, do not allocate
;   cmp y0,a y:MaxPos,r7       ; current sub-band vs end sub-band
;                               ; & set addr adj to max allocation
;   jlt <_bita_180_A           ; if passed, do not allocate
;   bset #ALLOCATE_SINE,x:(r6+n6) ; flag sub-band as a sine wave
;   jmp <_bita_110_A           ; if in range, allocate the maximum

```

\_bita\_97\_A

; otherwise,  
; find Signal-to-Noise position that puts Signal below Masking Threshold

```

;   move x1,r7                ; start at 1st Signal-to-Noise
;   position                  ;
;   move #SNR,n7              ; addr of Signal-to-Noise table
;   move x:(r1+n1),y0         ; get signal to mask ratio

```

```

;   dc #NUMSNRPOSITIONS-1,_bita_110_A

```

```

;   move y:(r7+n7),a          ; get the Signal-Noise at position
;   add y0,a                  ; add MNR to SNR for test
;   jle <_bita_100_A          ; still above mask, try next position

```

; now below the Global Mask, quit the loop

```

        enddo                                ;found position, stop #NUMSNRPOS-1
loop
    jmp <_bita_110_A                        ;go to end of loop

_bita_100_A
; try the next position and continue the loop

    move (r7)-                               ;try next Sig-Noise position

_bita_110_A                                ;END of #NUMSNRPOSITIONS-1 do loop

    move r7,y0                                ;save the matched SNR position
    move y:MaxPos,a                          ;to test if exceeded max position
    cmp y0,a y1,r3                          ;is counted pos greater than max
                                           ; & start at index 0 with allocation
    jge <_bita_115_A                        ;if not, go on to match the index
    move a1,y0                              ;set position at the maximum

_bita_115_A

;find index of the position that best matches the selected SNR
position

    do #NUMINDEXES,_bita_130_A

        move x:(r3+n3),a                    ;get the sub-band indexed position
        cmp y0,a                            ;compare to selected position
        jlt <_bita_120_A                    ;match not found yet, try next index

;found the matching index, quit the loop

        enddo                                ;found index, stop #NUMINDEXES loop

; a. if doing a sine wave in this sub-band, accept maximum
position index
; b. otherwise, see if maximum position assigned and if so,
; back up one index to the next to last index for this sub-band

        jset #ALLOCATE_SINE,x:(r6+n6),_bita_130_A ;if sine, accept
index
        move y:MaxPos,y0                    ;max position for Allowed table
selected
        cmp y0,a                            ;see if max position assigned
        jlt <_bita_130_A                    ;if not, accept the assigned index
        move (r3)-                          ;back up to the next-to-last index
        move x:(r3+n3),a                    ;assign the next-to-last index
position
        jmp <_bita_130_A                    ;go to end of loop

_bita_120_A

;try the next index and continue the loop

```

95

BAD ORIGINAL

```

        move r3 -                ;try position at next index
;see if end of the table line reached

        move x:(r3+n3),a        ;get this next index to test
        tst a                   ;test for an index of zero
        jne <_bita_125_A        ;if not 0, keep looking

;index of zero indicates no higher indices apply, back up 1 and use
that

        move (r3)-              ;use previous index
        bset #ALLOCATE_LIMIT,x:(r6+n6) ;set the completely
allocated bit
        bset #HEARING_LIMIT,x:(r6+n6) ;set the completely
allocated bit
        move x:(r3+n3),a        ;assign the last index position
        enddo                   ;found index, stop #NUMINDEXES loop
        jmp <_bita_130_A        ;go to end of loop

_bita_125_A
        nop                     ;keep looping

_bita_130_A                      ;END of #NUMINDEXES do loop

;set the initial allocation SubBandIndex and SubBandPosition

        move r3,x:(r5+n5)        ;set initial allocation SBIndx
        move a1,x:(r4+n4)        ;set initial allocation SBPos.

;determine the number of scale factor bits allocated at this
position

        move x:(r2+n2),n7        ;get the SBits scale factor code
(0-3)
        move #NSKFBits,r7        ;addr SBits scale factor bit count
tbl
        nop
        move y:(r7+n7),y0        ;save the scale factor bit count

;if joint stereo and we have reached the intensity sub-band
boundary
;    add the right channel joint SBits bit count also

        jclr #JOINT_at_SB_BOUND,y:<stereo,_bita_140_A
        move #NUMSUBBANDS,n2      ;offset to right channel Joint
SBits
        nop
        move x:(r2+n2),n7        ;get the SBits scale factor code
(0-3)
        move #0,n2               ;restore to left channel Joint SBits
        move y:(r7+n7),a        ;save the scale factor bit count
        add y0,a                ;add left to right Joint SBits cnt
        move a,y0               ;restore to proper register

```

\_bita\_140\_A

;add the bits required for the signal data

```

    move x:(r4+n4),n7      ;get the position
    move y:ndatabit,r7     ;addr of NDataBit count by position
    nop
    move y:(r7+n7),a       ;get the bit count at this position
    add y0,a y:TotBits,x0   ;add scale factor bits
                           ; & get curr TotBits
    add x0,a y:AvlBits,x0   ;update TotBits with bits just
allocated
                           ; & get available bits
    move a,y:TotBits        ;save new allocated total bits

```

; if joint stereo run at full, see if total available bits exceeded

```

    jclr #JOINT_at_FULL,y:<stereo,_bita_150_A
    cmp x0,a               ;check if room for allocation
    jle <_bita_150_A       ;if room, continue

```

; not enough room for FULL stereo, we have to do Joint Stereo  
; if already joint was sensed, continue developing demand bit rate

```

    jset #0,y:<jntflag,_bita_150_A ;joint sensed before,
continue

```

;1st indication of joint:

```

; indicate we found joint is needed
; save the sub-band number at this point

```

```

    bset #0,y:<jntflag      ;indicate joint sensed
    move y:count,a          ;get the sub-band number
    move a,y:jntsub         ;save the sub-band number for later

```

\_bita\_150\_A

;check that Signal-to-Noise position that Signal below Masking  
Threshold

```

    move x:(r4+n4),n7      ;get the position
    move #SNR,r7           ;addr of Signal-to-Noise table
    move x:(r1+n1),y0       ;get signal to mask ratio
    move y:(r7+n7),a        ;get the Signal-Noise at position
    add y0,a x:(r5+n5),r3   ;add MNR to SNR for test
                           ; & set up to set prev index for its pos
    jle <_bita_160_A        ;above mask, skip next statement
    bset #MASKING_LIMIT,x:(r6+n6) ;set AtLimit partially done
allocate

```

\_bita\_160\_A

;if joint stereo run at full, continue with the next channel

97

BAD ORIGINAL

```

;set #JOINT_at_FULL,y:<stereo,_bita_200_A
;if a sine wave sub-band, fill out total allocation
;set #ALLOCATE_SINE,x:(r6+n6),_bita_185_A

;for others,
;set the value for testing the best sub-band to deallocate bits
from
;if the frame cannot handle the full required allocation

    move (r3)-                ;back up one index to get that
position
    move x:(r3+n3),n7          ;get the position at the previous
index
    nop
    move y:(r7+n7),a           ;get the Signal-Noise at position
    add y0,a                   ;calc Sig-to-Noise at prev position
    move a,x:(r0+n0)           ;save in SBMNRmax array for later
    jmp <_bita_200_A           ;continue with the next channel

_bita_180_A

;if channel has a sine wave, suppress any allocation during final
passes

    move y:strtsin,a           ;get start sub-band if sine wave
    tst a #>1,y0               ;if -1, no sine wave
                                ; & set up to test sub-band 1, if sine
    jlt <_bita_185_A           ;if -1 not sine wave, continue

;for current sub-bands 0 or 1 to suppress any allocation

    move y:count,b             ;get current sub-band (00-31)
    tst b                      ;check if sub-band 0, to suppress alloc
    jeq <_bita_185_A           ;if 0, do not allocate
    cmp y0,b                   ;if sub-band 1, no allocate
    jeq <_bita_185_A           ;if 0, do not allocate

;for 1st harmonic sub-bands (start and end times 2) to suppress any
allocate
;all other sub-bands are set as at masking limit to allow some
allocation
;of leftover bits

    asl a                      ;double start subband suppress harmonic
    cmp a,b y:endsin,a         ;see if current sub-band harmonic
                                ; & get set to test end subband harmonic
    jeq <_bita_185_A           ;if harmonic, NO allocate
    asl a                      ;double end subband suppress harmonic
    cmp a,b                    ;see if current sub-band harmonic
    jne <_bita_190_A           ;set as at masking limit
    jeq <_bita_185_A           ;if harmonic, NO allocate

```



\_bita\_185\_A

;sub-band (channel) is not to be coded at all

bset #ALLOCATE\_LIMIT,x:(r6-n6) ;set AtLimit totally out of allocation

bset #HEARING\_LIMIT,x:(r6-n6) ;set AtLimit at threshold of hearing

\_bita\_190\_A

;sub-band (channel) is set to indicate it is at its masking threshold

bset #MASKING\_LIMIT,x:(r6+n6) ;set AtLimit partially done allocate

\_bita\_200\_A

;finished the sub-band at the current channel

; a. if just finished the right, skip next instructions

jclr #LEFT\_vs\_RIGHT,y:<stereo,\_bita\_210\_A

enddo ;to save cycles, stop #NUMCHANNELS

loop

jmp <\_bita\_220\_A

; b. otherwise, set up for the right

; set the left vs right channel flag indicating

; that right channel in process

; set the array register offsets to 32 sub-bands

\_bita\_210\_A

;initialize for the possible presence of a sine wave in right channel

;get the right xpsycho sine wave sub-bands to handle possible sine wave

move y:strtsinrgt,a ;start entry equals 1st sub-band

move a,y:strtsin ;isolate the starting sub-band

move y:endsinrgt,a ;end entry equals last sub-band

move a,y:endsin ;isolate the ending sub-band

bset #LEFT\_vs\_RIGHT,y:<stereo ;flag for right channel in progress

move #NUMSUBBANDS,n0 ;offset to the right channel SBMNRmax

move n0,n1 ;offset to the right channel SBMSr

move n0,n2 ;offset to right chan SBits

move n0,n6 ;offset to right chan AtLimit

move n0,n4 ;offset to right chan SBPos

move n0,n5 ;offset to right chan SBIndx



```

                                ;END of #NUMCHANNELS do loop
_bita_230_A
;set up for the initial allocation of the next subband

    move (r0)-                ;next sub-band SBMNRmax
    move (r1)-                ;next sub-band SBMSr
    move #16,r3                ;to position to next Allowed sb table
    move (r2)-                ;next sub-band SBbits or JointSbits
    move r3,-n3                ;next sub-band Allowed table array
    move r3,n3                ;set addr for next sub-band Allowed

pos
    move (r4)-                ;next sub-band SBPos
    move (r5)-                ;next sub-band SBIndx
    move y:count,r7            ;get current sub-band count
    move (r6)+                ;next sub-band AtLimit
    move r6,y:svereg           ;save updated AtLimit register
    move y:UsedSBReg,r6        ;get set to increment used counter

addr
    move (r7)+                ;increment the sub-band counter
    move (r6)+                ;next sub-band UsedSBs
    move r7,y:count            ;save new sub-band
    move r6,y:UsedSBReg        ;set incremented used counter addr
    move y:svereg,r6           ;restore AtLimit register

_bita_230_A                    ;END of #NUMSUBBANDS do loop

; if joint stereo does NOT apply, continue
    jclr #JOINT_FRAMING,y:<stereo,_bita_990_A

; if 2nd pass at Joint Stereo just completed, continue
    jset #JOINT_at_SB_BOUND,y:<stereo,_bita_990_A1

;if just finished the initial pass for JOINT stereo at FULL stereo
; if frame could not handle full stereo, set up the joint

    jset #0,y:<jntflag,_bita_235_A
    jclr #JOINT_at_FULL,y:<stereo,_bita_235_A

;the frame can handle FULL stereo, see if the previous frame
exhausted the
; continuous joint boundary frame counter

    move y:jfrmcnt,a           ;frame decrement count at last
boundary
    move #>1,x0                ;to decrement frame count at last
bound
    sub x0,a                   ;decrement the joint frame counter
    move a,y:jfrmcnt           ;save new joint frame counter
    nop
    tst a    #>FULL_STEREO,x1 ;see if frame count down over
                                ; & in case, set frame ISO stereo code
    jgt <_bita_235_A           ;if joint, use last frame's sub-band

```

```

cnt
;since the frame can handle FULL stereo, change the op frame type

    clr  a      x1,y:opfirtyp    ;to clear the history boundary
                                ; & set output frame as full stereo
    move a,y:boundlst            ;clear joint history sub-band
boundary
    move a,y:jfrmcnt             ;clear joint frame counter
    jmp  <_bita_990_A

_bita_235_A

;Joint at FULL stereo not possible, prepare for Joint Stereo
framing
;    store the demand rate

    bclr #JOINT_at_FULL,y:<stereo ;clear flag FULL not possible
    move y:fixbits,x0             ;get the constant bit count
    move y:TotBits,a              ;get bits required for frame
    add  x0,a                      ;set demand bits required
    move a,y:demand               ;save demand bit rate
    move y:frmtyp,x1              ;output frame as joint stereo
    move x1,y:opfirtyp            ;set new output frame type=JOINT

;do the joint calculation routines and prepare the proper arrays

    move #>BOUND_4,x1             ;default to lowest boundary
    move x1,y:sibound             ;set sub-band boundary for jointval
    move #polydta,r0              ;addr of left channel poly samples
    move #polydta,r1              ;to set addr of right channel
    move #INPCM,n1                ;offset to right channel poly samples
    move #JntPlAnal,r2            ;joint channel poly samples
    move (r1)+n1                  ;addr of right channel poly samples
    move #JntSBSKF,r3             ;addr of sub-band scale factors:
                                ; the joint left and right scale
                                ; factors
    move #JntSBMaxi,r4            ;joint channel Maxi factors by
                                ; sub-band and block of 12 samples
    jsr jointval                  ;calculate joint array values

;set the intensity sub-band boundary

    move y:jntsub,a               ;get sub-band where not at Mask
Thresh
    move y:q_psych,x0             ;get joint sub-band adjustment
    add  x0,a                      ;adjust joint sub-band count

;based on some pre-determined minimum joint sub-band,
; see if the sub-band count is to be forced to a higher value

    cmp  b,a                      ;count vs pre-set minimum sub-band
    jge  <_bita_236_A             ;if count above minimum, continue

```

```

;sub-band count is below the pre-determined joint sub-band
    move b,a                ;use pre-set minimum sub-band as count

_bita_236_A
    move #>BOUND_16,x1      ;start at highest boundary
    cmp x1,a #>INTENSITY_16,y0 ;test limit vs sub-band
                                ; & get frame header boundary code
    jge <_bita_240_A        ;we found the boundary
    move #>BOUND_12,x1      ;try the next highest boundary
    cmp x1,a #>INTENSITY_12,y0 ;test limit vs sub-band
                                ; & get frame header boundary code
    jge <_bita_240_A        ;we found the boundary
    move #>BOUND_8,x1       ;try the next highest boundary
    cmp x1,a #>INTENSITY_8,y0 ;test limit vs sub-band
                                ; & get frame header boundary code
    jge <_bita_240_A        ;we found the boundary
    move #>BOUND_4,x1       ;defaults to the lowest boundary
    move #>INTENSITY_4,y0   ;defaults to the lowest boundary

_bita_240_A
;test history of joint framing looking for a change in boundary

    mov y:boundlst,a        ;get current boundary
    tst a y:jfrmcnt,b       ;see if set previously
                                ; & get frame decr counter
    jle <_bita_242_AA        ;if not set, start new boundary &
count
;see if the frame decrement counter at zero

    tst b                   ;see if zero (or less)
    jle <_bita_242_AA        ;if done, start new boundary and
count
;compare last boundary to one just determined:
; if less, start with new higher boundary and restart the frame
decrement count
; if equal, continue without decrement frame counter
; else, decrement frame counter and switch to saved boundary and
ISO code

    cmp x1,a y:jfrmcnt,r0    ;compare boundaries
                                ; & get curr decr frame count
    jlt <_bita_242_AA        ;if less, start with new higher bound
    jeq <_bita_248_AA        ;if equal, continue

;since new frame has boundary less than history boundary:
; decrement frame counter
; use history boundary
; use history ISO code for the frame header

    move (r0)-               ;decrement the frame counter

```

```

        move y:boundlst,x1          ;switch to history intensity boundary
        move y:isocdelst,y0        ;switch to history boundary ISO code
        jmp <_bita_248_AA

_bita_242_AA
;start new history at current frame's intensity boundary and
;restart frame count

        move y:jntfrms,r0          ;initialize frame decrement count

_bita_248_AA
;set the frame header stereo intensity code

        move x1,y:sibound          ;set the sub-band boundary value
        move y0,y:stintns          ;for setsyst routine

;save current intensity boundary controls for the next frame

        move x1,y:boundlst         ;set last intensity boundary
        move y0,y:isocdelst        ;save ISO frame header code last used
        move r0,y:jfrmcnt          ;save frame decrement counter

;since doing joint stereo,
;pick correct joint scale factors for left channel then the right
channel
;first, see if testing with pickskf or pickjskf based on the factor
;applied to the demand bit rate with result compared to actual bit
rate

        bclr #1,y:<jntflag          ;use pickskf as default
        move y:p_psych,x1          ;get demand factor against demand
        move y:demand,x0           ;get the demand rate bit count
        mpy x0,x1,a y:AvlBits,x0   ;apply factor to demand bits
                                     ; & get available bits

;if demand rate * factor gives a result still greater than the
actual bit rate,
; use pickskf because bits are at a premium, otherwise, use
pickjskf

        cmp x0,a                  ;see adjusted demand still higher
        jge <_bita_242_A           ;if still higher, use pickskf
        bset #1,y:<jntflag         ;use pickjskf

_bita_242_A
        move #JntSBSKF,r0          ;addr of sub-band jnt scale
factors-left:
        move #JntSBits,r1          ;addr of jnt SBits array-left channel
        jclr #1,y:<jntflag,_bita_243_A
;!!!dbg
; nop
; nop

```



```

; nop
; nop
; nop
;!!!!dbg
;jsr pickjskf ;pick joint skf's for coding-left chan
;jmp <_bita_244_A

_bita_243_A
;!!!!dbg
; nop
; nop
; nop
; nop
; nop
;!!!!dbg
;jsr pickskf ;pick scale factors for coding-left chan
;!!!!tst jsr pickjskf ;pick joint skf's for coding-left
chan

_bita_244_A
;move #JntSBSKF,r0 ;addr of sub-band jnt scale
factors-left
;move #NUMSUBBANDS*NPERGROUP,n0 ;for right channel SKFs
offset
;move #JntSBits,r1 ;addr of jnt SBits array-left channel
;move #NUMSUBBANDS,n1 ;for right channel SBits offset
;move (r0)+n0 ;adjust for the start of right chan
SKFs
;move (r1)+n1 ;adjust for start of right chan SBits

;see if testing with pickskf or pickjskf

;jclr #1,y:<jntflag,_bita_246_A
;!!!!dbg
; nop
; nop
; nop
; nop
; nop
;!!!!dbg
;jsr pickjskf ;pick joint skf's for coding-right chan
;jmp <_bita_248_A

_bita_246_A
;!!!!dbg
; nop
; nop
; nop
; nop
; nop
;!!!!dbg
;jsr pickskf ;pick scale factors for coding-right chan
;!!!!tst jsr pickjskf ;pick joint skf's for coding-right
chan

```

109

BAD ORIGINAL

```

_bita_248_A
; determine the joint stereo bits available for bit allocation

    bclr #JOINT_at_FULL,y:<stereo ;now handle as joint stereo
    jsr bitpool ;set more available bits
    move x1,y:AvlBits

; restore original array of used sub-band count down counters for
; joint allocate

    move #UsedSBs,r0
    move #SvUsedSBs,r1
    do #NUMSUBBANDS*2,_bita_249_A
    move x:(r1)+,x0
    move x0,x:(r0)+

_bita_249_A
    jmp <_bita_40_A ;go back & redo the initial
allocation

_bita_990_A
; if not joint stereo, store the demand rate

    move y:fixbits,x0 ;get the constant bit count
    move y:TotBits,a ;get bits required for frame
    add x0,a ;set demand bits required
    move a,y:demand ;save demand bit rate

_bita_990_A1
; done with the initial allocation phase, phase A
; set the de-allocation passes initial state of control flags

    bset #MASKING_PASS,y:<stereo ;flag do masking passes
    bclr #HEARING_PASS,y:<stereo ;allocate index must be >
1
    bclr #FINAL_PASS,y:<stereo ;NOT final passes

; see if frame fits or do we have to de-allocate selectively

    move y:TotBits,x0 ;get the total bits allocated
    move y:AvlBits,a ;get available bits
    cmp x0,a ;TotBits vs BitsAvailable
    jge <_bita_990_B ;it fits, allocate any leftover bits

    do #1000,_bita_990_B

; test the bit allocation timeout flag
; if the timer flag was trip, switch over to the final bit
allocation
; of any remaining bits

```

105

BAD ORIGINAL

```

;clr #0,y:<global,_bita_10_B
;set #FINAL_PASS,y:<stereo,_bita_10_B ;continue, if final
;bset #FINAL_PASS,y:<stereo ;set for FINAL criteria

ON_BITALLOC_LED_CD ;tickle the led
enddo ;stop the #1000 loop and exit
move y:TotBits,x0 ;get the total bits allocated
jmp <_bita_990_C ;out of time, de-alloc under last
basis

_bita_10_B

;now let's look for qualifying candidates for next de-allocation
; we'll check out the left channel 1st, then the right

bclr #LEFT_vs_RIGHT,y:<stereo ;flag for left channel in
progress
move #SBMNRmax,r0 ;addr of de-alloc Max signal-noise
move y:BinXAdd,r5 ;set register of SubBandIndex
array
move #AtLimit,r6 ;point to SubBandAtLimit array
move #0,n0 ;offset to the left channel SBMNRmax
move n0,n5 ;offset to left chan SBIndx
move n0,n6 ;offset to left chan AtLimit
move #0,r2 ;use r2 as a sub-band counter
move r2,y:<MNRsub ;start cnt of de-allocate table
entries
move #>1,x1 ;to test for index of 1
move y:uselmsb,y1 ;to test for at least one alloc limit
move #MNRval,n3 ;get address of MNRval table
move #MNRsbc,n4 ;get address of MNRsbc table

;to deallocate the 1 index if the signal starts out below global
mask
move #SBMSr,r1 ;addr of Mask-to-Signal by sub-band
move n0,n1 ;offset to left chan SBMSr
move y:sibound,x0 ;joint stereo intensity sub-band
move x0,y:bandcnt ;bound subband decremented cnt
bclr #JOINT_at_SB_BOUND,y:<stereo ;clear reached boundary
sub-band

_bita_20_B

;loop thru the sub-bands for the current channel (left is 1st,
then, right)

do y:<usedsb,_bita_80_B

;to deallocate the 1 index if the signal starts out below global
mask
jclr #JOINT_FRAMING,y:<stereo,_bita_21_B
jset #JOINT_at_FULL,y:<stereo,_bita_21_B

```

106

BAD ORIGINAL



```

;set #JOINT_at_SB_BOUND,y:<stereo,_bita_21_B
move r3,y:svereg ;save reg 3
move y:bandcnt,r3 ;get decrement sub-band ctr
jsr chkjoint ;see if reached boundary
move r3,y:bandcnt ;save new decremented ctr
move y:svereg,r3 ;restore the saved reg 3
jclr #JOINT_at_SB_BOUND,y:<stereo,_bita_21_B
move #NUMSUBBANDS*2,n1 ;Joint SBMSr values by sub-band

_bita_21_B

;if no index has been allocated, try the next sub-band

move x:(r5+n5),a ;check for an allocated index
tst a ;if zero, try the next sub-band
jeq <_bita_70_B ;no allocation try next sub-band

;if a sine wave sub-band, do not deallocate

jset #ALLOCATE_SINE,x:(r6+n6),_bita_70_B

;if the 3rd mode of selection, no checks are made

jset #FINAL_PASS,y:<stereo,_bita_60_B ;3rd mode, use this
one

;if 2nd mode of selection sub-band may be below the masking
threshold, but
; checks to make sure that if index allocated is ONE and that
the
; sub-band is not required for continuity

jset #HEARING_PASS,y:<stereo,_bita_50_B ;2nd mode num of index

;must be 1st mode of selection which requires that the sub-band
; be below the masking threshold

jclr #MASKING_LIMIT,x:(r6+n6),_bita_70_B ;skip: above mask
thresh

_bita_50_B

;if we have allocated only 1 index, skip this sub-band if at least
one
; allocation is required

cmp x1,a ;see if index at 1
jgt <_bita_60_B ;no, this sub-band qualifies

;to deallocate the 1 index if the signal starts out below global
mask

move r2,a ;get current sub-band
cmp y1,a ;see if sub-band below at least 1

```

```

; if greater, deallocation candidate
; if greater than 14, check
; test sb vs 14
; & restore uselmsb to y1
; if less than 14, keep the 1
allocation
    move x: r1-n1, b ; get Max Signal to MinMask
    tst b ; if positive, started below global mask
    jle <_bita_70_B ; if not positive, keep the 1
allocation
    _bita_60_B
; candidate qualifies,
; insert this candidate into the table for initial de-allocation
    jsr insert_value

    _bita_70_B
; advance to the next sub-band for the current channel

    move (r2)+ ; increment the sub-band counter
    move (r0)+ ; next sub-band SBMNRmax
    move (r5)+ ; next sub-band SBIndx
    move (r6)+ ; next sub-band AtLimit

    _bita_80_B ; end of y:kusedsb do loop

; if we just finished the right channel,
; let's see if we have any candidates to de-allocate
    jset #LEFT_vs_RIGHT, y: < stereo, _bita_90_B

; let's go thru the right channel looking for de-allocation
candidates
    bset #LEFT_vs_RIGHT, y: < stereo ; flag for right channel in
progress
    move #SBMNRmax, r0 ; addr of de-alloc Max signal-noise
    move y: BInxAdd, r5 ; set register of SubBandIndex
array
    move #AtLimit, r6 ; point to SubBandAtLimit array
    move #NUMSUBBANDS, n0 ; offset to the right channel
SBMNRmax
    move n0, n5 ; offset to right chan SBIndx
    move n0, n6 ; offset to right chan AtLimit
    move #0, r2 ; use r2 as a sub-band counter
    move #MNRsbc, n4 ; get address of MNRsbc table

; to deallocate the 1 index if the signal starts out below global
mask
    move n0, n1 ; offset to right chan SBMsr

```

```

        move y:sibound,x0          ;point stereo intensity sub-band
        move x0,y:siboundcnt      ;bound subband decremented cnt
        bclr #COUNT_at_SB_BOUND,y;<stereo ;clear reached boundary
sub-band

        jmp <_bita_20_B          ;no look thru right channel sub-bands
_bita_90_B

;if there are any entries in the de-allocate tables, start
reclaiming bits

        move y:<MNRsub,a          ;get the de-allocate table entry cnt
        tst a                    ;test for zero, no entries
        jne <_bita_110_B         ;are entries at this criteria,
dealloc

;since there were no candidates to deallocate (MNRsub = 0),
; change the selection criteria:
;   if we've done the final criteria and nothing to de-allocate,
;       we can do nothing here, exit (How Come???)
;   if we've not found anything with at least 2 indexes allocated,

;       switch to select from any sub-bands
;   if we've not found anything below the masking threshold,
;       switch to at least 2 indexes alloc
;redo the selection criteria

        jset #FINAL_PASS,y:<stereo,_bita_092_B
        jset #HEARING_PASS,y:<stereo,_bita_100_B
        jset #MASKING_PASS,y:<stereo,_bita_105_B
        bset #MASKING_PASS,y:<stereo
        jmp <_bita_200_B         ;loop thru with this criteria
_bita_092_B

;see if a sine wave in either or both channels and if so open them
up for
;deallocation

        move #AtLimit,r6         ;address of AtLimit array both
channels
        jset #LEFT_SINE_WAVE,y:<stereo,_bita_94_B
        jset #RIGHT_SINE_WAVE,y:<stereo,_bita_96_B

;if no sine wave and still too much????? shouldn't be, exit

        enddo                    ;stop the #1000 loop and exit
        move y:TotBits,x0        ;get the total bits allocated
        jmp <_bita_990_C

_bita_94_B

;clear the sine wave indicators from left channel and open up for

```

deallocation

```

        move y:strtssinlft,n6          ;1st sine wave sub-band of
adjacent pair
        bclr #LEFT_SINE_WAVE,y:<stereo    ;clear the indicator
        bclr #ALLOCATE_SINE,x:(r6+n6)    ;clear the hold allocate if sine
        move y:endsinlft,n6            ;2nd sine wave sub-band of adjacent
pair
        nop
        bclr #ALLOCATE_SINE,x:(r6+n6)    ;clear the hold allocate if sine
        jclr #RIGHT_SINE_WAVE,y:<stereo,_bita_200_B ;loop with this
criteria

```

\_bita\_96\_B

;clear the sine wave indicators from right channel and open up for deallocation

```

        move #NUMSUBBANDS,n6            ;offset to right channel
        nop
        move (r6)+n6                    ;shift over to right channel flags
        move y:strtssinrgt,n6          ;1st sine wave sub-band of
adjacent pair
        bclr #RIGHT_SINE_WAVE,y:<stereo    ;clear the indicator
        bclr #ALLOCATE_SINE,x:(r6+n6)    ;clear the hold allocate if sine
        move y:endsinrgt,n6            ;2nd sine wave sub-band of adjacent
pair
        nop
        bclr #ALLOCATE_SINE,x:(r6+n6)    ;clear the hold allocate if sine
        jmp <_bita_200_B                ;loop thru with this criteria

```

\_bita\_100\_B

```

        bclr #HEARING_PASS,y:<stereo
        bset #FINAL_PASS,y:<stereo
        jmp <_bita_200_B                ;loop thru with this criteria

```

\_bita\_105\_B

```

        bclr #MASKING_PASS,y:<stereo
        bset #HEARING_PASS,y:<stereo
        jmp <_bita_200_B                ;loop thru with this criteria

```

;there are entries in the de-allocate tables

\_bita\_110\_B

;de-allocate from the table from 1st entry to last  
; or until enough bits have been reclaimed

```

        clr a
        move a,y:count                ;start counter thru the table

;loop through the ordered de-allocation table

        dc y:<MNRsub,_bita_190_B

```

110

BAD ORIGINAL

```

        move #MNRsb, r0          ;address of MNRsb table
        move y:count, r0         ;current table entry index
        bclr #LEFT_vs_RIGHT, y: <stereo ;default to left channel
        bclr #JOINT_at_SB_BOUND, y: <stereo ;clear reached boundary
sub-band
        move x: (r0+n0), a        ;get selected sub-band/channel
        move a, y: <MNRsb         ;isolate selected sub-band/channel
        move r0, -                ;increment to next table entry
        move r0, y:count         ;save next table entry
        bclr #6, y: <MNRsb, _bita_130_B ;if left channel, continue
        bclr #6, y: <MNRsb         ;right channel sub-band number
        move y: <MNRsb, a         ;get corrected sub-band (0-31)
        bset #LEFT_vs_RIGHT, y: <stereo ;indicate right channel

_bita_120_B
;restore the left channel array addresses
        move #SBMNRmax, r0        ;addr of de-alloc Max signal-noise
        move #SBMSr, r1           ;addr of Mask-to-Signal by sub-band
        move y:BitsAdd, r2        ;set register of SBits array

;if doing a Joint Stereo frame (not upgraded to FULL),
; if the sub-band is included in the intensity coding,
;   set the SBMSr and SBits to the joint arrays
        jclr #JOINT_FRAMING, y: <stereo, _bita_130_B ;NOT Joint
framing
        jset #JOINT_at_FULL, y: <stereo, _bita_130_B ;Joint upgraded
to FULL

;compare the selected sub-band to the stereo intensity sub-band
limit
;if not at or above the limit, continue as normal
;   otherwise switch to Joint array addresses
        move y: sibound, x0        ;get intensity sub-band limit
        cmp x0, a                 ;compare the two
        jlt <_bita_130_B          ;we're doing a stereo sub-band

;we're at intensity sub-band limit
; shift over to Joint channel in SBMSr array (3rd set of sub-band
values in n1)
; and the Joint SBits
        bset #JOINT_at_SB_BOUND, y: <stereo ;set reached boundary
sub-band
        move #NUMSUBBANDS*2, n1    ;Joint SBMSr values by sub-band
        nop
        move (r1)-n1               ;adjust addr to JointSBMSr
        move #JntSBits, r2         ;set register of JointSBits
array
        _bita_130_B

```

111

BAD ORIGINAL



```

;continue restoring the left channel array addresses

        move    y:SBPosAdd,r4        ;set register of SubBandPosition
array    move    y:SBInxAdd,r5        ;set register of SubBandIndex
array    move    #AtLimit,r6         ;point to SubBandAtLimit array

;if from left channel, addresses are OK. otherwise, offset for
right channel

        jolt #LEFT_vs_RIGHT.y:<stereo,_bita_140_B

        move    #NUMSUBBANDS,n0        ;offset to the right channel
SBMNRmax
        move    n0,n1                ;offset to the right channel SBMSr
        move    n0,n2                ;offset to right chan SBits
        move    n0,n4                ;offset to right chan SBPos
        move    n0,n5                ;offset to right chan SBInx
        move    n0,n6                ;offset to right chan AtLimit
        move    (r0)+n0              ;offset register for SBMNRmax to
right
        move    (r1)+n1              ;offset register for SBMSr to right
        move    (r2)+n2              ;offset register for SBbits to right
        move    (r4)+n4              ;offset register for SBPos to right
        move    (r5)+n5              ;offset register for SBInx to right
        move    (r6)+n6              ;offset register for AtLimit to right

_bita_140_B

;set the proper allowed table of indexed position based on the
selected sub-band

        move    y:AllwAdd,r3        ;init the current Allowed table
        tst     a                    ;see if it's sub-band zero (from above)
        jeq     <_bita_150_B        ;sub-band zero was selected
        move    #16,n3                ;to increment to next sub-band addr
        do      a,_bita_150_B        ;increment to sub-band number chosen
        move    (r3)+n3              ;16 position entries per sub-band

_bita_150_B
        move    r3,n3                ;set Allowed addr for sub-band chosen
        move    y:<MNRsb,n0          ;get selected sub-band in SBMNRmax
        move    n0,n1                ;sub-band in SBMSr
        move    n0,n2                ;sub-band in SBbits
        move    n0,n4                ;sub-band in SBPos
        move    n0,n5                ;sub-band in SBInx
        move    n0,n6                ;sub-band in AtLimit
        move    y:ndatabit,r7        ;addr of NDataBit count by position
        move    y:TotBits,a          ;get current bits allocated
        move    x:(r5+n5),r3         ;get the current allocated index
        move    x:(r4+n4),n7         ;get the position at the old index
        move    (r3)-                ;back up one index
        move    r3,x:(r5+n5)         ;save new SBInx for sub-band

```

112

BAD ORIGINAL

```

    move y:(r7-n7),x0      ;data bits allocated at that position
    sub x0,a               ;subtract old allocated data bits
    move x:(r3-n3),n7      ;get new position
    move n7,x:(r4-n4)      ;save new SBPos for sub-band
    move y:(r7-n7),b       ;data bits allocated at new position
    add b,a                ;add new allocated data bits

    tst b                  ;see if index 1 just de-allocated
    jne <_bita_160_B       ;if not, save the new TotBits value

;we have to take off the scale factor bits

    move x:(r2-n2),n7      ;get the SBits scale factor code
(0-3)
    move #NSKFBits,r7      ;addr SBits scale factor bit count
tbl
    nop
    move y:(r7+n7),y0      ;get the scale factor bit count
    sub y0,a               ;subtract from TotBits

;if joint stereo and we have reached the intensity sub-band
boundary
; add the right channel joint SBits bit count also for this
sub-band

    jclr #JOINT_at_SB_BOUND,y:<stereo,_bita_160_B
    move #>NUMSUBBANDS,b   ;offset to right channel Joint
SBits
    move n2,x0             ;sub-band
    add x0,b               ;offset to right channel subband
    move b1,n2             ;access right channel-sband Joint
SBits
    nop
    move x:(r2+n2),n7      ;get the SBits scale factor code
(0-3)
    nop
    move y:(r7+n7),y0      ;save the scale factor bit count
    sub y0,a               ;subtract from TotBits

_bita_160_B
    move a,y:TotBits       ;save the new total bits

;check if Signal-to-Noise position that Signal above/below Masking
Threshold

    bclr #MASKING_LIMIT,x:(r6+n6) ;clear AtLimit below masking
threshold
    move x:(r4+n4),n7      ;get the position
    move #SNR,r7           ;addr of Signal-to-Noise table
    move x:(r1-n1),y0      ;get signal to mask ratio
    move y:(r7+n7),a       ;get the Signal-Noise at position
    add y0,a x:(r5+n5),r3  ;add MNR to SNR for test
    ; & set up to set prev index for its pos
    jle <_bita_170_B       ;above mask, skip next statement

```

```

        cset #MASKING_LIMIT,x: r6-n6      ;set AtLimit below masking
threshold
        _bita_170_B

;check if the bit pool can now handle the frame as allocated

        move y:TotBits,a                ;get the new total bits
        move y:AvlBits,x0              ;get the available bits
        cmp x0,a                        ;BitsAvailable vs TotBits
        jgt <_bita_180_B                ;need more, continue with
de-allocation

        enddo                          ;we're done here, stop MNRsub loop
        enddo                          ;we're done here, stop #1000 loop
        jmp <_bita_990_B

        _bita_180_B

;if there is no index allocated (r3 = 0), continue with the next
table entry

        move r3,a                      ;get newly decremented index allocated
        tst a (r3)-                    ;if it is zero, continue
        jeq <_bita_185_B                ; & back up one index for that position
        ;allocated index equals 0, continue

;set the value for testing the best sub-band to deallocate bits
from
;if the frame cannot handle the full required allocation

        move x:(r3+n3),n7              ;get the position at the previous
index
        nop
        move y:(r7+n7),a                ;get the Signal-Noise at position
        add y0,a                        ;calc Sig-to-Noise at prev position
        move a,x:(r0+n0)                ;save in SBMNRmax array for later

        _bita_185_B                    ;continue y:<MNRsub do loop
        nop

        _bita_190_B                    ;end of y:<MNRsub do loop
        nop

        _bita_200_B                    ;continue #1000 do loop
        nop

        _bita_990_B                    ;end of #1000 dc loop

; set the allocation passes initial state of control flags

        bset #MASKING_PASS,y:<stereo    ;flag do masking passes
        bclr #HEARING_PASS,y:<stereo    ;NOT hearing threshold
passes

```



```

        bclr #FINAL_PASS,y:<stereo      ;NOT final passes

;get the total bits allocated so far
        move    y:TotBits,x0

; Now that we have the initial bit allocation, iterate on it.
; for LoopCount = 1; --LoopCount
do      #1000,_bita_990_C

;test the bit allocation timeout flag
; if the timer flag was trip, switch over to the final bit
allocation
; of any remaining bits

        jclr #0,y:<qtalloc,_bita_10_C
        jset #FINAL_PASS,y:<stereo,_bita_10_C
        bset #FINAL_PASS,y:<stereo

; ON_BITALLOC_LED_CD      ;tickle the led

;this is equivalent to the call to the c subroutine:
;
;(c) AllocateBits()
;
;initial allocation is done, set-up for as needed allocation loop
;restore the left channel array addresses

_bita_10_C
        move    #SBMSr,r1      ;set register of SBMSr array
        move    y:BitsAdd,r2   ;set register of SBits array
        move    y:BPosAdd,r4   ;set register of SubBandPosition
array
        move    y:BInxAdd,r5   ;set register of SubBandIndex
array
        move    #AtLimit,r6    ;point to SubBandAtLimit array
;
;(c)      FirstTime = 1;      /*start run thru subbands this time
*/
;
        bset #FIRST_TIME,y:<stereo      ;FirstTime = !0

        clr     a
        move    a1,y:count      ;start the sub-band counter
        move    y:AllwAdd,r0     ;init the current Allowed table
        move    #SNR,r3

;in case of joint stereo, clear the reached intensity sub-band
boundary flag

        move    y:sibound,x1     ;joint stereo intensity sub-band
        move    x1,y:bandcnt     ;bound subband decremented cntr

```

115

BAD ORIGINAL



```

        bclr #JOINT_at_SB_BOUND,y:<stereo ;clear reached boundary
sub-band

;go through all used sub-bands for both channels looking at only
those
; that have not reached the allocation limit

        do          y:<usedsb,_bita_130_C

;clear the n registers for the left channel reference

        clr a
        move a,n1          ;SBMSr array
        move a,n2          ;SBits array
        move a,n4          ;SBPos array
        move a,n5          ;SBIndx array
        move a,n6          ;AtLimit array

;clear the left vs right channel flag indicating that left channel
in process

        bclr #LEFT_vs_RIGHT,y:<stereo ;flag for left channel in
progress

; if joint stereo does NOT apply, continue

        jclr #JOINT_FRAMING,y:<stereo,_bita_30_C

; if joint stereo upgraded to full, continue

        jset #JOINT_at_FULL,y:<stereo,_bita_30_C

; if doing joint stereo and have already switched over to joint
SBits array,
; but now have to adjust to 3rd set of SBMSr values

        jset #JOINT_at_SB_BOUND,y:<stereo,_bita_20_C

; see if the joint stereo intensity sub-boundary has been reached
; if not, continue at full stereo for these early sub-bands
; otherwise, switch over to the JointSBits and Joint channel in
; the SBMSr array (3rd set of sub-band values (n1

        move r3,y:svereg          ;save reg 3
        move y:bandcnt,r3          ;get decrement sub-band ctr
        jsr chkjoint              ;see if reached boundary
        move r3,y:bandcnt          ;save new decremented ctr
        move y:svereg,r3          ;restore the saved reg 3
        jclr #JOINT_at_SB_BOUND,y:<stereo,_bita_30_C

        move #JntSBits,r2          ;shift over to Joint SBits array
        move y:count,n2            ;to offset to current sub-band
        nop
        move (r2)+n2              ;adj addr to current sub-band

```

```

        move #0,n2                ;reset to left channel

_bita_30_C

;we're at intensity sub-band limit
; shift over to Joint channel in SBMSr array
; 3rd set of sub-band values in n1

        move #NUMSUBBANDS*3,n1    ;Joint SBMSr values by sub-band

_bita_30_C

;process the current channel

        do #NUMCHANNELS,_bita_120_C

;see if this sub-band's limit flag was set previously, and skip if
it has
;
;(c)      if( Left(or Right)AtLimit[SubBand] )
;(c)      continue;
;
        jset #ALLOCATE_LIMIT,x:(r6+n6),_bita_100_C ;skip subband
reached limit

        jset #FINAL_PASS,y:<stereo,_bita_40_C ;pass skips below mask
check

        jset #MASKING_LIMIT,x:(r6+n6),_bita_100_C ;skip subband
reached limit

_bita_40_C
        move x:(r4+n4),a          ;get curr position[SubBand]

;see if this sub-band has reached its limit already
;
;(c)      if( Left(or Right)SubBandPosition[SubBand] == MaxPos ) {
;(c)      Left(or Right)AtLimit = 1;
;(c)      continue;
;(c)      }
;
        move y:MaxPos,y0          ;set max position
        cmp y0,a a1,n3            ;see if max position
; & move pos to n3
        jeq _bita_80_C            ;reached its allocation limit,
set flag

;check this sub-band out
; see if there is room to handle the next allocation for this
sub-band
;
;(c)      NextSubBandPosition =
;(c)      AllowedPositions[SubBand]
;(c)      (Left(orRight)SubBandIndex[SubBand]+1);

```



```

; c      TestBits = OldTotBits
; c      - NDataBits[Left or Right SubBandPosition[SubBand]]
; c      - NDataBits[NextSubBandPosition];
; c      if Left or Right SubBandIndex[SubBand] == 0
; c      TestBits += NSKFBits0Left or Right SBits[SubBand];
;
; c      clr b      #>1,y1      ;init added scale factor bits
; c      ; & to incr to next allowed bits size
; c      move x:(r5-n5),a      ;SubBandIndex[SubBand]

; if this will be the 1st index, we must account for the scale
; factor bits

; c      tst a      #NSKFBits,r7      ;see if 0
; c      ; & set addr of NSKFBits array
; c      jne <_bita_50_C      ;not 1st index, skip add scale bits

; set the scale factor - sbits needed for this 1st index in this
; sub-band

; c      move x:(r2+n2),n7      ;get SBits index
; c      nop
; c      move y:(r7+n7),b      ;num bits for scaling info

; if joint stereo and we have reached the intensity sub-band
; boundary
; c      add the right channel joint SBits bit count also

; c      jclr #JOINT_at_SB_BOUND,y:<stereo,_bita_50_C
; c      move #NUMSUBBANDS,n2      ;offset to right channel Joint
; c      SBits
; c      nop
; c      move x:(r2+n2),n7      ;get the SBits scale factor code
; c      (0-3)
; c      move #0,n2      ;restore to left channel Joint SBits
; c      move y:(r7+n7),y0      ;save the scale factor bit count
; c      add y0,b      ;add left to right Joint SBits cnt

; c      _bita_50_C
; c      add y1,a y:ndatabit,r7      ;increment
; c      ; & addr of NDataBit count by position
; c      move a1,n0      ;set offset for Allowed next index

; do not allocate position 17, stop this sub-band if that is the
; case

; c      move #>17,y1      ;get the biggest position to test
; c      move x:(r0+n0),a      ;get the NextPosition as the new pos
; c      cmp y1,a      ;see if biggest allocation
; c      jeq <_bita_80_C      ;do not, end allocation this sub-band

; see if next allocation is passed the max for this sub-band as per
; Allowed table
; c      this has happened if this next position is zero

```

```

        tst     a     a1,n7          ;see if passed the maximum position
        ; & move new pos to n7
        reg     <_bita_80_C         ;reached its allocation limit,
set flag

;test the allocation at this new position

        move    y:(r7+n7),y1        ;get NDataBits[NextSBPos]
        add     y1,b n3,n7          ;add to any scaling info bits
        ; & set offset SubBandPos[SubBand]
        move    b1,y1              ;bits to add for next index
        move    x0,b               ;b==>TestBits = OldTotBits
        move    y:(r7+n7),y0        ;get NDataBits[SBPos[SubBand]]

        sub     y0,b a1,x1          ;TestBits -= current bits
        ; & put new position in proper reg
        add     y1,b y:AvlBits,a    ; TestBits += next allocation bits
        ; & gets BitsAvailable

;
;(c)         if( TestBits > BitsAvailable ) {
;(c)             Left(or Right)AtLimit = 1;
;(c)             continue;
;(c)         }
;
        cmp     b,a b,y:TotBits     ;see if room & save allocation
        jlt     <_bita_80_C         ; no room, set as AtLimit and
continue

;if this is the final loop, skip the next test and allocate the
bits

        jset    #FINAL_PASS,y:<stereo,_bita_70_C ;pass skips below mask
check

;
;(c)         SMR = Left(or Right)SubBandMax[SubBand]
;(c)             - Left(or Right)MinMaskingDb[SubBand]
;(c)         MNR = SNR[Left(or Right)SubBandPosition[SubBand]] - SMR
;
        move    y:(r3+n3),y1        ;get SNR[SubBandPos[SubBand]]
        move    x:(r1+n1),a         ;SBMSr[SubBand] Mask-to-Signal
        add     y1,a y:MNRmin,b     ;add Sig-Noise ratio;
        ; & get MNRmin for below
        jgt     <_bita_90_C         ;below Masking, go to take out
partially

;
;(c)         if( FirstTime || MNR < MNRmin ) {
;(c)             MNRsb = SubBand;
;(c)             MNRchan = channel;
;(c)             MNRmin = MNR;
;(c)             FirstTime = 0;
;(c)         }
;
        move    a,y1                ;save MNR

```

```

;set #FIRST_TIME,y:<stereo,_bita_60_C ;if first, save as
minimum
cmp     y1,b ;MNRmin = MNR
;le     _bita_100_C

_bita_60_C
move n0,y:MNRinx ;MNRinx = NewIndex;
move x1,y:MNRpos ;MNRpos = NewPosition;
move y:TotBits,x1 ;get the allocation of bits
move x1,y:HldBits ;save the allocation of bits
move y:count,x1 ;get current sub-band
move x1,y:<MNRsb ;MNRsb = SubBand;
move n2,y:MNRchan ;MNRchan = 0 if left, 32 if right
move y1,y:MNRmin ;MNRmin = MNR;
bclr #FIRST_TIME,y:<stereo ;clear FirstTime flag

jmp     _bita_100_C

;we are on the final allocations passes after all channels for all
sub-bands
; are driven below the Global Masking threshold

_bita_70_C
move y:TotBits,x0 ;save new TotBits
move n0,x:(r5+n5) ;save new sub-band index
move x1,x:(r4+n4) ;save new allocation position
bclr #FIRST_TIME,y:<stereo ;clear FirstTime flag
jmp <_bita_100_C

_bita_80_C
bset #ALLOCATE_LIMIT,x:(r6+n6) ;set the completely
allocated bit
bset #HEARING_LIMIT,x:(r6+n6) ;set the completely
allocated bit

_bita_90_C
bset #MASKING_LIMIT,x:(r6+n6) ;set the reached global
masking bit

;finished the sub-band at the current channel,
; a. if just finished the right, skip next instructions

_bita_100_C
jclr #LEFT_vs_RIGHT,y:<stereo,_bita_110_C
enddo
jmp <_bita_120_C

; b. otherwise, set up for the right
; set the left vs right channel flag indicating
; that right channel in process
; set the array register offsets to 32 sub-bands

_bita_110_C
bset #LEFT_vs_RIGHT,y:<stereo ;flag for right channel in

```

```

progress
    move #NUMSUBBANDBS,n1          ;offset to the right channel
SBMSr
    move n1,n2                    ;offset to right chan SBits
    move n1,n6                    ;offset to right chan AtLimit
    move n1,n4                    ;offset to right chan SBPos
    move n1,n5                    ;offset to right chan SBIndx

;finished both channels for this sub-band, now set up for the next
subband

_bita_120_C
    move y:count,r7              ;get current sub-band to increment
    move #16,n0                  ;now update Allowed to next
sub_band
    move (r1)+                   ;SBMSr array
    move (r2)+                   ;SBits array
    move (r4)+                   ;SBPos array
    move (r5)+                   ;SBIndx array
    move (r6)+                   ;AtLimit array
    move (r0)+n0                 ;advance Allowed to next
sub-band
    move (r7)+                   ;increment the sub-band counter
    move r7,y:count              ;save new sub-band number

_bita_130_C
; At this point the following registers are in use
;   y:AvlBits = # of bits available
;   y:<MNRsb = MNRsb
;   y:MNRMin = MNRmin
;
;We test now to see if this trip thru the loop produced any changes
; and if not, we have finished the bit allocation for this frame.
;
;(c) if( FirstTime )
;(c)     return;
;
;   jclr #FIRST_TIME,y:<stereo,_bita_140_C ;not 1st, alloc
to selected
;   jclr #FINAL_PASS,y:<stereo,_bita_160_C ;not final, set
1 more loop
;finished, end the loop and go to exit routine

enddo
jmp <_bita_990_C

_bita_140_C
;test flag all candidates are below masking threshold

jset #FINAL_PASS,y:<stereo,_bita_170_C ;if final,
allocated already

```

; restore the left channel array addresses

```

        move    y:SBPosAdd,r4        ;set register of SubBandPosition
array   move    y:SBInxAdd,r5        ;set register of SubBandIndex
array   move    y:MNRchan,a          ;get indication as to which channel
        tst     a                    ;0 if from left channel
        beq     <_bita_150_C        ;if from left channel, addrs OK
        move    #NUMSUBBANDS,n4      ;offset to right SBPos channels
        move    n4,n5                ;offset to right SBInx channels
        move    (r4)+n4              ;offset register for SBPos to right
        move    (r5)+n5              ;offset register for SBInx to right

```

\_bita\_150\_C

```

;       SubBandIndex[MNRsb]++
;       SubBandPosition[MNRsb] =
AllowedPositions[MNRsb][SubBandIndex[MNRsb]]

```

```

        move    y:<MNRsb,n5          ;MNRsb
        move    n5,n4                ;MNRsb
        move    y:MNRinx,x1          ;get the saved new index
        move    x1,x:(r5+n5)         ;update the SBInx for selected
sub-band
        move    y:MNRpos,x1          ;get the saved new allowed position
        move    x1,x:(r4+n4)         ;update the SBPos for selected
sub-band
        move    y:HldBits,x0         ;set the new bit allocation total cnt
        jmp     <_bita_170_C         ;continue major loop

```

; now lets just allocate what's left now that all are below mask

```

_bita_160_C
        bset    #FINAL_PASS,y:<stereo ; just loop now

```

```

_bita_170_C
        nop

```

\_bita\_990\_C

; restore the register 7 values

```

        move    x:bitallocM7Save,m7
        move    x:bitallocN7Save,n7
        move    x:bitallocR7Save,r7

```

```

;       bclr    WATCH_DOG           ;tickle the dog
;       nop

```

```

; check for any sub-band with no index allocation
; and if zero, zero it's in use count down counter

```

```

        move    y:SBInxAdd,r5        ;addr allocated indexes

```

122

BAD ORIGINAL



```

        move #UsedSSBs,r6          ;addr of used sub-band counters
        dc   =NUMSUBBANDS*2,_bita_994_C

        move x,r5--,a              ;get allocated index
        tst  a                     ;see if zero
        jne  <_bita_992_C          ;if not zero, continue
        move a,x:r5               ;reset count down counter to zero

_bita_993_C
        move (r6)-                 ;incr addr to next counter

_bita_994_C

;set up the padded bits count for ancillary data

        move x0,y:TotBits          ;save bits actually allocated
        move y:AvlBits,b           ;determine number of bits padded
        sub  x0,b                  ;bits available minus total allocated
        move b1,y:padbits          ;save count of unallocated audio bits
        rts

;insert_value():
;
;This routine orders the table of values per sub-band/channel
;that are to be de-allocated as needed. The table is ordered in
;descending sequence that makes the 1st entry the one that can best
;afford a deallocation.

;on entry:
;
;    x:(r0+n0) = the current value to be inserted
;    y:<stereo = bit 1 indicates left channel (0) or right channel
;(1)
;    r2 = the sub-band number to be inserted
;    y:<MNRsub = current count of entries in the ordered
deallocation tables
;    n3 = address of MNRval table
;    n4 = address of MNRsbc table
;
;on exit:
;
;    y:<MNRsub = incremented count of entries in ordered
deallocation tables
;
;    a = destroyed
;    b = destroyed
;    x0 = destroyed
;    y0 = destroyed
;    r3 = destroyed
;    r4 = destroyed

        org  phe:
        org  pli:

```

insert\_value

;get the current value to be inserted and set up the start into  
; the ordered table of values and the associated table of  
sub-band/channels

move x:(r3+n0),a ;get the current value to insert  
move y:<MNRsub,b ;get current count of table entries

;if this is the 1st value to be inserted into the table, skip the  
; search for its place and enter this as table entry no 1

tst b #0,r3 ;see if this is 1st entry into table  
; & set to 1st entry in MNRval table  
jeq <\_insert\_50 ;if 1st, skip following table search

;search through the table of entries so far established looking for  
where  
;to store this current value

do y:<MNRsub,\_insert\_20

move x:(r3+n3),x0 ;get the table value for comparison  
cmp x0,a ;against the new value to be inserted  
jlt <\_insert\_10 ;if less, value is further down table

;when the new value is greater than or equal to the table entry,  
; this is its place in the table, we may have to shift the  
following  
; table entries in order to enter this new value

enddo ;stop the y:<MNRsub do loop  
jmp <\_insert\_20 ;see if the table must be shifted

\_insert\_10  
move (r3)+ ;try the next table entry

\_insert\_20 ;end of y:<MNRsub do loop

;if this entry number (its place in the table) equals the count of  
entries,  
; this entry will be the new LAST entry in the table

move r3,x0 ;get its place in the table to  
compare  
cmp x0,b ;its place to current table entry count  
jgt <\_insert\_25 ;if less, we have to shift the table  
jeq <\_insert\_50 ;if eq, entry is appended to the  
table  
move b1,r3 ;?? let's make sure we use last entry  
jmp <\_insert\_50

\_insert\_25

124

BAD ORIGINAL

;we need to shift the subsequent entries in the table down one and  
 then  
 ; insert this new sub-band/channel value

```

    move b1,r3                ;establish the curr table ends
    move b1,r4                ;for both MNRval and MNRsbc
    move (r3)+n3              ;set r3 with addr of MNRval end - 1
    move (r4)+n4              ;set r4 with addr of MNRsbc end - 1
    move r3,-                 ;back off 1 to get last MNRval entry
    sub x0,b (r4)-            ;number of table entries to shift
                                ; & back off 1 to get last MNRsbc entry

    do b,_insert_40           ;shift each down 1 position in tables

    move x:(r3)+,y0            ;get curr value and incr to rec addr
    move y0,x:(r3)-            ;put value 1 entry down & back up 1
    move x:(r4)+,y0            ;curr sub-band/chan & incr to rec
addr
    move y0,x:(r4)-            ;put value 1 entry down & back up 1
    move (r3)-                 ;back up one more entry table MNRval
    move (r4)-                 ;back up one more entry table MNRsbc

    _insert_40                ; end of b do loop

```

;restore entry location to receive value and sub-band/channel

```

    move x0,r3

```

```

    _insert_50

```

;insert the current value at this location in the ordered table  
 ; also insert the sub-band number and set the channel flag

```

    move r3,r4                ;matching position in the MNRsbc
table
    move a,x:(r3+n3)          ;enter sorted value
    move r2,x:(r4+n4)          ;enter the sub-band number
    jclr #LEFT_vs_RIGHT,y:<stereo,_insert_99
    bset #6,x:(r4+n4)          ;flag as the right channel

```

```

    _insert_99

```

;increment the count of entries in the ordered deallocation tables

```

    move y:<MNRsub,r3          ;we need to increment entry counter
    nop
    move (r3)+
    move r3,y:<MNRsub          ;save the new table entry count

    rts

```

125

BAD ORIGINAL

opt      fs,mex

© 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.

UXCODE\rcode.asm: the ULTIMA cdq2000 with 2 xpsycho's and xcode in one DSP  
 updated to support 48000, 44100, 32000, 24000, 22050 and 16000 sampling rates

title    'MUSICAM Transitter Main'  
 stitle   'Initialization'

include 'def.asm'  
 include '..\common\ioequ.asm'  
 include 'box\_ctl.asm'  
 include 'box\_tbls.asm'  
 include 'translte.asm'

page

;In a given MUSICAM frame time period this routine performs the XPSYCHO  
 ;function on both channels followed by the XCODE functions of bit  
 ;allocation and frame encoding.

```

section lowmisc
xdef    word_out
xdef    word_in1
xdef    word_in2
xdef    word_in3
xdef    not_appl
xdef    starty
xdef    maxsubs
xdef    maxcritbnds
xdef    ipwptr
xdef    frmbits
xdef    fixbits
xdef    audbits
xdef    usedsb
xdef    stereo
xdef    cmprsc1
xdef    sibound
xdef    nmskfregs
xdef    cutmus
xdef    outsize
xdef    timer
xdef    timeout
xdef    qtalloc
xdef    oprptr
xdef    frmstrt
xdef    frmnext
xdef    plctmn
xdef    plct11
xdef    plct12
xdef    dbgcnt
xdef    endy

xdef    limitsb

org      yli:
stxcode_yli

```

126

BAD ORIGINAL

```

word_out      ds      1      ;applicable output (leds, switches)
word_in1      ds      1      ;applicable input (switches, lines)
word_in2      ds      1      ;applicable input (switches, lines)
word_in3      ds      1      ;applicable input (switches, lines)
    _appl      ds      1      ;satisfy non-applicable hardware settings

starty

maxsubs       ds      1      ;working MAXSUBBANDS for sample/bit rates
maxcritbnds   ds      1      ;MAXCRITBNDs for sample/bit rates
ipwptr        ds      1      ;input PCM buffer write pointer (even = left)
frmbits       ds      1      ;bits in the audio portion of frame
fixbits       ds      1      ;bits required before audio data bits
audbits       ds      1      ;number of bits available for audio data
usedsb        ds      1      ;number of used sub-bands
stereo        ds      1      ;y:<stereo = flags:
                    ;bit 0 means stereo vs mono framing
                    ; 0 = stereo framing
                    ; 1 = mono framing
                    ;bit 1 indicates left vs right channel
                    ; 0 = looping thru left channel arrays
                    ; 1 = looping thru right channel arrays
                    ;bit 2 indicates joint stereo applies
                    ; 0 = NOT joint stereo framing type
                    ; 1 = IS joint stereo framing type
                    ;bit 3 indicates curr frame upgraded to
                    ; full stereo by joint bit allocation
                    ; (if joint stereo applies)
                    ; 0 = normal joint stereo allocation
                    ; 1 = FULL STEREO allocation
                    ;bit 4 indicates the stereo intensity
                    ; sub-band boundary has been reached
                    ; (if joint stereo applies)
                    ; 0 = NO sub-bands still below
                    ; intensity boundary
                    ; 1 = sub-bands above intensity
                    ; boundary
                    ;bit 5 is FirstTime switch in a loop
                    ; thru the bit allocation
                    ; 0 = cleared if any allocations
                    ; were made
                    ; 1 = no allocations made to any
                    ; sub-band
                    ;bit 6 indicates a below masking
                    ; threshold allocation pass
                    ; 0 = some sub-bands not below mask
                    ; 1 = all sub-bands are below mask
                    ;bit 7 indicates a below hearing
                    ; threshold allocation pass
                    ; 0 = some sub-bands not below hearing
                    ; threshold
                    ; 1 = all sub-bands are below hearing
                    ; threshold
                    ;bit 8 indicates final bit allocation
                    ; passes to use up any available bits
                    ; 0 = not yet
                    ; 1 = allocate remainder in bit pool
                    ;bit 9 indicates limit of sub-bands requiring
                    ; at least one position has been reached:
                    ; 0 = not yet, 1 = limit reached

```

127

BAD ORIGINAL

```

;bit 10 indicates maximum limit of sub-bands
; that are to be allocated has been reached:
; 0 = not yet, 1 = limit reached
;bit 11 indicates whether or not dual
; transmission output lines apply and
; that the block sequence number must be
; appended to the frame
; 0 = NO block sequence number
; 1 = YES append the block sequence number
;bit 12 indicates that the split framing mode
; applies (go to MONO if one line is down)
; 0 = split framing does not apply
; 1 = split framing does apply
;bit 13 indicates to do a split mono frame
; because one line is down
; 0 = code a normal frame
; 1 = code a split mono frame

cmprscctl ds 1 ;control flags for CCS compression:
; bit 0 = application:
; 0 = ISO standard
; 1 = CCS compression applies

sibound ds 1 ;intensity subband boundary alloc
nmskfreg ds 1 ;NMSKFREQS for sample/bit rates
outmus ds 1 ;number of words to read in
outsize ds 1 ;circular buffer ctl frame o/p buffer
timer dc 0 ;frame sync timer interrupt sensor:
; bit 0 set by irqb - received frame sync
; bit 1 after 1st frame skipped if sync failure

meout dc 0 ;frame sync failure counter
alloc ds 1 ;frame msec timer interrupt bit alloc
; signal bit allocator to finish up

oprptr ds 1 ;read pointer into frame buffer
frmstrt ds 1 ;starting addr of current frame
frmnext ds 1 ;start addr of frame 2 to align with frame sync
plctmn ds 1 ;successive phase lock detect high conter main
plctl1 ds 1 ;successive phase lock detect high conter line 1
plctl2 ds 1 ;successive phase lock detect high conter line 2
dbgcnt ds 1 ;!!! debug counter

endy

limitsb dc 0 ;LIMITSUBBANDS ;sub-bands req at least 1 allocation

endxcode_yli
endsec

```

```

section lowmisc
xdef startx_xli
xdef polyst
xdef rtonals
xdef nmasker
xdef nalislft
xdef nalisrgt
xdef maxtonal
xdef maxbin
xdef sinbin
xdef sincnt
xdef sintest
xdef SvReg0

```

128

BAD ORIGINAL

```

        xdef      dbptr
        xdef      endx_xli

        org      xli:
        rtx_xli
        polyst      ds      1      ;addr of the polyanalysis start
        ntonals      ds      1      ;number of tonals in tonal structure
        nmasker      ds      1      ;number of maskers in masker structure
        nalislft      ds      1      ;number aliasers - left channel
        nalisrgt      ds      1      ;number aliasers - right channel
        maxtonal      ds      1      ;to see if sine wave, highest tonal
        maxbin      ds      1      ;if sine wave, bin num of highest tonal
        sinbin      ds      1      ;bin number of sine wave must persist
        sincnt      ds      1      ;frame cnt to see if sine wave persists
        sintest      ds      1      ;channel tester to see if sine wave
        SvReg0      ds      1      ;Save Register 0
        dbptr      ds      1      ;!!!debug
        endx_xli

```

endsec

section highmisc

```

        xdef      bitrate
        xdef      frmrate
        xdef      rawrate
        xdef      smplrte
        xdef      smplcde
        xdef      smplidbit
        xdef      padrate
        xdef      paddiff
        xdef      padrest
        xdef      usediff
        xdef      bndwidth
        xdef      frmtype
        xdef      opfrtyp
        xdef      maxsubbands
        xdef      stintns
        xdef      oldccs
        xdef      strtstnlft
        xdef      endstnlft
        xdef      strtstnrgt
        xdef      endstnrgt
        xdef      sincntlft
        xdef      sincntrgt
        xdef      sintstlft
        xdef      sintstrgt
        xdef      rngtbl
        xdef      xaxisincr
        xdef      thresh
        xdef      thresslb
        xdef      holdthresslb
        xdef      splitthresslb
        xdef      b_i
        xdef      fmap
        xdef      cb
        xdef      g_cb
        xdef      dbaddtbl
        xdef      curxlft
        xdef      curxrgt
        xdef      frmformat

```

```

xdef reedsolomon
xdef trailbits
xdef reedendpos
xdef psychaddr
xdef dbgaddr
xdef dbgflag

```

tables of variables for sampling rate, framing bit rate and baud rate

```

xdef samplerates
xdef translaterates
xdef bitrates
xdef framevalues
xdef psychtable
xdef bauddata

```

org yhe:

stxcode\_yhe

```

bitrate      ds      1      ;ISO frame header bit rate code as per frmrate
frmrate      ds      1      ;frame bit rate index for table manipulation
                                ; for either high or low sampling rate:
                                ;   code      high sampling    low sampling
                                ;   0          384             160
                                ;   1          256             144
                                ;   2          192             128
                                ;   3          128             112
                                ;   4          112             96
                                ;   5           96             80
                                ;   6           64             64
                                ;   7           56             56
                                ;   8          320             48
                                ;   9          224             40
                                ;  10          160             32
                                ;  11           80             24
                                ;  12           48             16
                                ;  13           32              8
                                ;  14 (free)   399             399
rawrate      ds      1      ;raw input frame bit rate to be translated
                                ; switches (5 bits) indicate
                                ; 00000 = 384 Kbits
                                ; 00001 = 256 Kbits
                                ; 00010 = 192 Kbits
                                ; 00011 = 128 Kbits
                                ; 00100 = 112 Kbits
                                ; 00101 = 96 Kbits
                                ; 00110 = 64 Kbits
                                ; 00111 = 56 Kbits
                                ; 01000 = 320 Kbits
                                ; 01001 = 224 Kbits
                                ; 01010 = 160 Kbits
                                ; 01011 = 80 Kbits
                                ; 01100 = 48 Kbits
                                ; 01101 = 32 Kbits
                                ; 01110 = 144 Kbits
                                ; 01111 = 40 Kbits
                                ; 10000 = 24 Kbits
                                ; 10001 = 16 Kbits
                                ; 10010 = 8 Kbits

```



smplrte	ds	1	; 10011 = 399 Kbits (free bit rate)
smplcde	ds	1	; audio sampling bit rate as to hardware
			; ISO frame hdr sample rate code as per smplrte
			; switches (2 bits) indicate
			; 00 = 44.1 K or 22.05 K
			; 01 = 48 K or 24 K
			; 10 = 32 K or 16 K
			; 11 = CDQ1000 mono at 24 K sampling
smplidbit	ds	1	; hdr id bit:
			; 1 for 44.1, 48 and 32 K sample rates
			; 0 for 22.05, 24 and 16 K sample rates
padrate	ds	1	; frame padding calculation: sample rate
paddiff	ds	1	; frame padding calc: DIFF @ sample/bit rates
padrest	ds	1	; frame padding calculation: REST
usediff	ds	1	; working diff for pad calculation
bndwidth	ds	1	; code for setting sub-band limits
frmtype	ds	1	; switches (2 bits) are set to:
			; 00 = (0) full stereo
			; 01 = (1) joint stereo
			; 10 = (2) dual channel
			; 11 = (3) mono (1 channel)
opfrtyp	ds	1	; current frame type after bit allocation
			; if unit coding joint stereo, the
			; frame could be full stereo as well
			; as joint stereo
maxsubbands	ds	1	; MAXSUBBANDS for sample/bit rates
stintns	ds	1	; intensity subband boundary code
oldccs	ds	1	; encode MPEG-ISO vs old CCS CDQ2000's
			; 0 = MPEG-ISO
			; 1 = old CCS CDQ2000's
ltsinlft	ds	1	; left channel -1 NOT sine, else 1st sub-band
endsinlft	ds	1	; left channel -1 NOT sine, else 2nd sub-band
strtsinrgt	ds	1	; right channel -1 NOT sine, else 1st sub-band
endsinrgt	ds	1	; right channel -1 NOT sine, else 2nd sub-band
sincntlft	ds	1	; sine test frame counter left channel
sincntrgt	ds	1	; sine test frame counter right channel
sintstlft	ds	1	; sine test flag left channel
sintstrgt	ds	1	; sine test flag right channel
rngtbl			; table for searching for tonals
dc			2,3,6,6,12,12,12,12
xaxisincr	ds	1	; x axis increment for b_i & ThresSLB tables
thresh	ds	1	; threshold of hearing table choice for XPSYCHO
thresslb	ds	1	; table address for current frame
holdthresslb	ds	1	; normal frames table addr for current frame
splitthresslb	ds	1	; mono split frames table addr for current frame
b_i	ds	1	; table address for current frame
fmap	ds	1	; table address for current frame
cb	ds	1	; table address for current frame
g_cb	ds	1	; table address for current frame
dbaddtbl	ds	1	; table address for current frame
curxlft	ds	1	; left channel-current location in x vector
curxrgt	ds	1	; right channel-current location in x vector
frmformat	ds	1	; communications frame formatting code

;Reed/Solomon frames controls:

reedsolomon	ds	1	; Reed/Solomon switch (bit 0) 0=no y=yes
trailbits	ds	1	; Reed/Solomon bits to take from end of frame
reedendpos	ds	1	; Reed/Solomon bit count - frame flush zero bits



```

psychaddr      ds      1      ;addr psychtable as per current sampling rate
dbgaddr        ds      1      ;!!! debug save address
dbgflag        dc      0      ;!!! debug control flag
stsmplrts_yhe

```

## SAMPLERATES

```

endsmplrts_yhe
sttransl_yhe

```

## TRANSLATERATES

```

endtransl_yhe
stbitrts_yhe

```

## BITRATES

```

endbitrts_yhe
stfrmvals_yhe

```

## FRAMEVALUES

```

endfrmvals_yhe
stpsychtbl_yhe

```

## PSYCHTABLE

```

endpsychtbl_yhe
stbauddata_yhe

```

## BAUDDATA

```

endbauddata_yhe

```

```

endxcode_yhe

```

```

endsec

```

```

section ptable

```

```

xdef      ptable
xdef      a_psych,b_psych
xdef      c_psych,d_psych
xdef      e_psych,f_psych,g_psych
xdef      h_psych,i_psych,j_psych
xdef      k_psych,l_psych,m_psych,n_psych,o_psych,p_psych
xdef      q_psych,r_psych,s_psych,t_psych,u_psych,v_psych,w_psych,x_psych
xdef      y_psych,z_psych
xdef      z1_psych,z2_psych,z3_psych,z4_psych,z5_psych,z6_psych

```

```

org      yhe:
stptable_yhe

```

```

ptable

```

```

;is table is loaded with IRT factors from 12/92

```

```

a_psych      dc      0.0467146      ;A curval
b_psych      dc      0.0498289      ;B curval
c_psych      dc      0.0259526      ;C curval

```

132

BAD ORIGINAL

```

d_psych      dc  0.04982389      ;D curval
e_psych      dc  0.0882387       ;E curval
f_psych      dc  0.40000000      ;F curval
g_psych      dc  0.0311431      ;G curval
h_psych      dc  0.0882387       ;H curval
i_psych      dc  0.0882387       ;I curval
j_psych      dc  0.10000000      ;J curval
k_psych      dc  0.00000000      ;K curval
l_psych      dc  0.00000000      ;L curval
m_psych      dc  0.00000000      ;M curval
n_psych      dc  0.00000000      ;N curval
o_psych      dc  0.00000000      ;O curval
p_psych      dc  0.00000000      ;P curval
q_psych      dc  0.00000000      ;Q curval
r_psych      dc  0.00000000      ;R curval
s_psych      dc  0.00000000      ;S curval
t_psych      dc  0.00000000      ;T curval
u_psych      dc  0.00000000      ;U curval
v_psych      dc  0.00000000      ;V curval
w_psych      dc  0.00000000      ;W curval
x_psych      dc  0.00000000      ;X curval
y_psych      dc  0.00000000      ;Y curval
z_psych      dc  0.00000000      ;Z curval
z1_psych     dc  0.00000000      ;Z1 curval
z2_psych     dc  0.00000000      ;Z2 curval
z3_psych     dc  0.00000000      ;Z3 curval
z4_psych     dc  0.00000000      ;Z4 curval
z5_psych     dc  0.00000000      ;Z5 curval
z6_psych     dc  0.00000000      ;Z6 curval

```

```

    .dptable_yhe
    endsec

```

```

    org     phe:
start

```

```

; The external wait state is set to 1. This allows the HCT541's to
; put their data on the bus in plenty of time.

```

```

    XCODE_M_BCR      ;set all external io wait states

```

```

;set new crystal clock to 64 MHz

```

```

    XCODE_M_PCTL

```

```

;load X and Y memory

```

```

    jsr     boot_xy

```

```

;!!!dbg jsr     initdeb ;!!!debug init the debug port
;!!!dbg move    #>S720906,a ;!!!debug
;!!!dbg jsr     outhex ;!!!debug
;!!!dbg jsr     cr ;!!!debug

```

```

; Clear all of the x memory

```

```

    clr     a ;value to set x memory to
    move    #ffff,m0 ;just in case, set to linear buffer
    move    #starty,r0 ;set starting address
    move    #endy-starty,r1 ;set loop count

```

```

        rep      r1                ;clear r1
        move     a,y:(r0)-
; set the start of the block sequencing table for the 1st frame
        move     =seqnums,r1
        move     r1,y:nxtseq
; PORT C Assignments
;
; s = ssi port
; i = input port
; o = output port
;
        XCODE_PORT_C_M_PCC          ;set port C control register
        XCODE_PORT_C_M_PCD          ;set output data to port C
        XCODE_PORT_C_M_PCDDR        ;set port C data direction reg
; initialize the ssi port for the ad converter
        XCODE_SSI_M_CRA              ;set ssi cra register
        XCODE_SSI_M_CRB              ;set ssi crb register
; initialize the sci port for tty
        XCODE_SCI_M_SCR              ;set sci status control register
; PORT B Assignments
;
; 14 13 12 - 11 10 9 8 - 7 6 5 4 - 3 2 1 0
;  o  o  i   o  i i o   o i i i   i i i i
;
        XCODE_PORT_B_M_PBC          ;set B control register for general IO
        XCODE_PORT_B_M_PBD          ;set the default outputs
        XCODE_PORT_B_M_PBDDR        ;set B register direction
; initialize the AES-EBU chip
        XPSYCHO_AES_EBU_INIT
; initialize the host vectors
        INIT_HOST_VECTORS_CD
restart
; set the interrupt for host interrupts
; HOST set to IPL 2
        movep    #>S0800,x:<<M_IPR    ;set int priorities and edges
        andi     #Sfc,mr              ;turn on the interrupt system
;
        ori      #S03,mr
        nop
        nop
        nop
; clear the analog to digital converter to restart calibration

```

```

        CLR_ADC_RESET
;disable the ancillary data received interrupt
        bclr    #M_RIE,x:<<M_SCR
;initialize the led applies word and light initial leds
        move    #>OFF_LEDS_CD,b          ;initialize leds as off
        move    b,y:<word_out
        ON_ALARM_LED_CD                    ;light alarm led indicator
        TST_SET_ALARM_RELAY_CD,_set_led_0 ;unless already set,
        SET_ALARM_RELAY_CD                ;set the alarm relay line on
_set_led_0
;see if an invalid bit rate was selected for the sampling rate
        OFF_INVALID_BIT_RATE_LED_CD       ;signal ok
        move    #smplidbit,r0             ;to test for bit rate error flag
        nop
        jclr    #4,y:(r0),_lite_leds      ;if no rate error, light the leds
;an invalid bit rate was selected for sampling rate
        ON_INVALID_BIT_RATE_LED_CD        ;signal the error
_lite_leds
        SET_LEDS_CD
;initialize the encoder control word: y:<stereo
        clr     a
        move    a,y:<stereo
;!!!dbg: initialize the debug counter
        move    a,y:dbgcnt
;!!!dbg
; init left and right channel start addresses in working x vector buffer
; and start initializing various registers for both channels
        move    #xbuflft,r0               ;left channel set start pos in x buffer
        move    r0,y:curxlft              ;save left channel start pos x buffer
        jsr     polyaini                  ;left channel init poly analysis filter
        move    #xbufrgt,r0               ;right channel set start pos in x buffer
        move    r0,y:curxrgt              ;save right channel start pos x buffer
        jsr     polyaini                  ;right channel init poly analysis filter
;initialize for joint framing intensity boundary control
        move    a,y:boundlst               ;zero last established boundary
        move    a,y:jfrmcnt               ;zero the frame count down ctl
;set up for receiving left and right channel PCM samples:
; left channel values are stored on even addresses in the buffer with

```

135

BAD ORIGINAL

```

; right channel values stored in the adjacent odd buffer addresses

move    #inpcm,r7                ;get the input pcm data buffer
move    r7,y:<lpwptr              ;set start left channel buffer address

;initialize for finding a sine wave as input

move     a,x:<sincnt               ;zero the sine frame counter
move     a,y:sincntlft            ;zero left channel sine frame counter
move     a,y:sincntrgt            ;zero right channel sine frame counter
move     a,x:<sintest              ;clear the sine indicator
move     a,y:sintstlft            ;clear the sine indicator left channel
move     a,y:sintstrgt            ;clear the sine indicator right channel

;initialize the array of sub-band usage

move     #UsedSBs,r0              ;addr of used sub-band counters
rep      #NUMSUBBANDS*NUMCHANNELS
move     a,x:(r0)+

;indicate this is the 1st frame after a restart for scale factor checksum

bclr     #2,x:private

;check the switches to determine bit rate and framing type
; and ancillary data application and data baud rate

GET_SWITCHES_CD gsws_00
jsr      getsws

move     x:tstrate,y1             ;set the frame rate i/p code
move     y1,y:rawrate
move     x:tstsmpl,y1             ;set the sampling rate i/p code
move     y1,y:smplrte
move     x:tstfrme,y1             ;stereo, mono or joint frames
move     y1,y:frmtype
move     x:tstband,y1             ;set allocation band width code
move     y1,y:bndwidth
move     x:tstbaud,y1             ;set ancillary data baud rate code
move     y1,y:baudrte
move     x:tstsel1,y1             ;set whether or not line 1 selected
move     y1,x:select1
move     x:tstsel2,y1             ;set whether or not line 2 selected
move     y1,x:select2
move     x:tstoccs,y1             ;set MPEG-ISO (0) or old CCS (1)
move     y1,y:oldccs
move     x:tstfrmt,y1             ;set the communication frame format
move     y1,y:frmformat
move     x:tstreed,y1             ;set Reed/Solomon frame format switch
move     y1,y:reedsolomon
;!!! move     x:tstbits,y1
;!!! move     y1,y:trailbits      ;bits taken from frame for Reed/Solomon

;set framing mode led

move     y:frmtype,a              ;get specified framing via switches
move     a,y:opfrtyp              ;set current frame type for output to
; the coded frame (this can change
; from frame to frame from JOINT_STEREO
; to FULL_STEREO if the JOINT_STEREO bit

```

136

BAD ORIGINAL

```

; allocation applies and can handle the
; curr frame data as true full stereo)

SET_FRAME_TYPE_LED_CD          ;light the proper leds

move    y:frmttype,a           ;get specified framing via switches
move    #>MONO,x0              ;start with mono
cmp     x0,a    #>JOINT_STEREO,x0 ;compare and set up for JOINT
jne     <_xcod_05

OFF_JOINT_LED_CD              ;clear the JOINT stereo led
OFF_STEREO_LED_CD            ;clear the FULL stereo led
ON_MONO_LED_CD               ;light the MONO led

;indicate mono framing (default is stereo)

bset    #STEREO_vs_MONO,y:<stereo

jmp     <_xcod_07

_xcod_05
cmp     x0,a                  ;if not JOINT, defaults to full stereo
jne     <_xcod_06

OFF_MONO_LED_CD              ;clear the MONO led
OFF_STEREO_LED_CD            ;clear the FULL stereo led
ON_JOINT_LED_CD              ;light the JOINT stereo led

;indicate joint stereo framing (default is not joint)

bset    #JOINT_FRAMING,y:<stereo

jmp     <_xcod_07

_xcod_06
OFF_MONO_LED_CD              ;clear the MONO led
OFF_JOINT_LED_CD             ;clear the JOINT stereo led
ON_STEREO_LED_CD             ;light the FULL stereo led

_xcod_07
SET_LEDS_CD

;based on the sampling rate and bit rate selected:
; set the frame header sampling rate code
; set the frame header bit rate code
; set the frame size in words and bits
; set the MAXSUBBANDS (for BALs)
; set the applicable bit allocation control parameters

move    #samplerates,r0      ;addr of sample rate values
move    #DATABYSAMPLERATE,n0 ;num parameters per sample rate
move    y:smpirte,b          ;to see if need to adjust address
tst     b                    ;if code 0, no need to shift address
jeq     <_smpirte            ;if 0, get the 3 parameters

;just the table address to proper sampling rate parameters

rep     b
move    .r0)+n0

```

```

_smplrte
    move    y:(r0)+,x0      ;get the ISO frame header ID code
    move    x0,y:smplidbit  ;save the ISO frame header ID code
    move    y:(r0)+,x0      ;get the ISO frame header code
    move    x0,y:smplcde     ;save the ISO frame header code
    move    y:(r0)+,x0      ;get the MAXCRITBnds for XPSYCHO
    move    x0,y:<maxcritbnds ;save the MAXCRITBnds for XPSYCHO
    move    y:(r0)+,x0      ;get the NMSKFREQS for XPSYCHO
    move    x0,y:<nmskfreqs   ;save the NMSKFREQS for XPSYCHO
    move    y:(r0)+,x0      ;get the sample rate value for pad calc
    move    x0,y:padrate     ;save sample rate value for pad calc
    move    y:(r0)+,x0      ;get value to determine b_ii & ThresSLB
    move    x0,y:xaxisincr   ;save value for b_ii & ThresSLB tbls

```

;translate the raw bit rate code to the internal index rate code  
 ; based on whether the sampling rate is high (y:smplidbit 1=high) or low (0)  
 ;and validate that the rate is supported by the software and/or hardware

```

    move    #translaterates,r0 ;addr of the translation table
    move    y:rawrate,n0        ;to offset to translated index
    bclr    #4,y:smplidbit      ;clear bad bit rate for sampling rate
    move    (r0)+n0             ;pos to bit rate translate 1st value
    move    (r0)+n0             ;pos to bit rate translate 2nd value
    move    y:smplidbit,n0      ;low (0) or high (1) sample rate select
    move    #>-1,a              ;to see if not supported
    move    y:(r0+n0),x0        ;get the translated rate index code
    cmp     x0,a                ;see if not supported rate
    jne     <_set_frmrate       ;if supported rate, set y:frmrate

```

ampling rate does not support the selected bit rate

```

    bset    #4,y:smplidbit
    jmp     restart

```

\_set\_frmrate

;set the framing bit rate table index code

```

    move    x0,y:frmrate

```

;position to the proper set of framing bit rate parameters  
 ; based on high or low sampling rate selected

```

    move    #smplidbit,r1      ;to test for high sampling rate
    move    #bitrates,r0       ;addr of framing bit rate parms
    move    #BITRATESLOWOFFSET,n0 ;in case of low sampling rate
    jset    #0,y:(r1),_get_ISO ;if high rate, continue
    move    (r0)+n0            ;position to low sampling values

```

\_get\_ISO

;get the framing bit rate ISO header code and possible split rate parameters

```

    move    #DATAByBITRATE,n0  ;num parameters per bit rate code
    move    y:frmrate,b        ;to see if need to adjust address
    tst     b                  ;if code 0, no need to shift address
    jeq     <_bitrate          ;if 0, get the 3 parameters

```

;adjust the table address to proper bit rate parameters



```

        rep      b
        move     (r0)+n0

bitrate
        move     y:(r0)+,x0          ;get the ISO frame header code
        move     x0,y:bitrate        ;save the ISO frame header code
        move     y:(r0)+,x0          ;get hi or lo rate threshold tbl ident
        move     x0,y:chresh         ;store hi or lo rate threshold tbl ident
        move     #0,n2               ;init as not split rate capable
        move     #0,r2               ;init as not split rate capable
        move     y:(r0)+,a           ;get optional split frame bit rate
        tst      a                   ;see if split rate applies
        jeq      <_setsplit         ;if not, set as null split rate parms

;indicate split frame mode of transmission can be used for this bit rate

        bset     #SPLIT_APPLIES,y:<stereo
        move     a,n2                 ;save optional split frame rate
        move     y:(r0)+,r2          ;get rate code for split rate band width

_setsplit

;set split rate parameters

        move     n2,y:spltrte        ;split mono frame rate code - header
        move     r2,y:spltbnd        ;split mono frame rate code - bandwidth

;get the framing parameters based on sampling rate and framing bit rate

        move     #framevalues,r0     ;addr of sample rate values
        move     #FRAMEBYSAMPLE,n0   ;numb parameters per sample rate
        move     y:smplrte,b         ;to see if need to adjust address
        tst      b                   ;if code 0, no need to shift address
        jeq      <_frbitrte         ;if 0, get the 3 parameters

;adjust the table address to proper sampling rate parameters

        rep      b
        move     (r0)+n0

_frbitrte
        move     #FRAMEBYBITRATE,n0  ;numb parameters per framing bit rate
        move     y:frmrate,b          ;test bit rate to set audio data size
        tst      b                   ;if code 0, no need to shift address
        jeq      <_frmdata          ;if 0, get the parameters

;adjust the table address to proper framing bit rate parameters at sample rate

        rep      b
        move     (r0)+n0

_frmdata
        move     y:(r0)+,x0          ;get the words per frame at rate
        move     x0,y:<outmus         ;save the words per frame at rate
        move     y:(r0)+,x0          ;get the bit count per frame at rate
        move     x0,y:<frmbits        ;save the bit count per frame at rate
        move     y:(r0)+,x0          ;get pad calc diff value @ samp/bit rtes
        move     x0,y:paddiff         ;save pad calc diff value
        move     x0,y:usediff         ;init as pad calc diff value
        move     (r0)+               ;step over bit offset unpadding

```

```

move    y:(r0)+,x0          ;get whether CCS compression applies
move    x0,y:<cmprscctl      ;set CCS compression (1) or not (0)
move    y:(r0)+,x0          ;get optional MAXSUBBANDS for split rate
move    x0,y:splitmaxsubs    ;split mono frame rate MAXSUBBANDS
move    y:(r0)+,x0          ;get split frame pad calc diff value
move    x0,y:splitpaddiff    ;save split frame pad calc diff value

;set MAXSUBBANDS based on one or two channels coded at this bit rate

move    #oldccs,r1          ;to test MPEG-ISO vs old CCS
move    #0,n0               ;set for two-channel MAXSUBBANDS
jset    #0,y:(r1),_old_ccs   ;if set, do CCS the old way (use MONO)
jclr    #STEREO_vs_MONO,y:<stereo,_maxsubs ;if 2 channel, offset is set

_old_ccs
move    #1,n0               ;set for one-channel MAXSUBBANDS
nop

_maxsubs
move    y:(r0+n0),x0        ;MAXSUBBANDS at rate and num channels
move    x0,y:maxsubbands    ;save the MAXSUBBANDS at this rate
move    x0,y:<maxsubs        ;set the working MAXSUBBANDS

;if old CCS switch is set.
; see if CDQ1000 old mono frames at 24 K sampling and bit rate of 56 or 64

move    #oldccs,r0          ;to see if old CCS requested
move    #>SAM24K,x0          ;to check on 24 K sampling
jclr    #0,y:(r0),_aft_old_CCS ;if not old CCS requested, continue
jclr    #STEREO_vs_MONO,y:<stereo,_aft_old_CCS ;if 2 channel, continue
move    y:smplrte,a          ;to test for 24 K sampling
cmp     x0,a    #>RATE56_LOW,x0 ;see if sampling at 24 K
jne     <_aft_old_CCS        ;if not continue
move    y:frmrate,a          ;to test the framing bit rate
move    #>BITRATE_56,x1      ;set high sampling rate bit rate code
cmp     x0,a    #>RATE64_LOW,x0 ;see if 56 K frame rate
jeq     <_set_cdq1000        ;if so, go to set up as if CDQ1000
cmp     x0,a    #>BITRATE_64,x1 ;see if 64 K frame rate
jne     <_aft_old_CCS        ;& set high sampling rate bit rate code
jne     <_aft_old_CCS        ;if not, continue

_set_cdq1000

;we are doing a CDQ1000 mono frame at 24 K sampling

move    #>SAMPLE_ID_BIT_HIGH,x0 ;set the frame header sampling id to 1
move    x0,y:smplidbit        ;to insert 1 in the frame header
move    #>SAMPLINGRATE_24_CDQ1K,x0 ;CDQ1000 sample rate code is 3
move    x0,y:smplcde          ;set code for setsyst rtn
move    #>27,x0               ;MAXSUBBANDS = 27
move    x0,y:maxsubbands      ;save the MAXSUBBANDS at this rate
move    x0,y:<maxsubs          ;set the working MAXSUBBANDS
move    x1,y:bitrate          ;set frame header bit rate code
bset    #0,y:<cmprscctl        ;do CCS compression

ft_old_CCS

;now set the type of ancillary count control:
; 0 = 3-bit pad byte count:
;      CDQ2000 @ 48 K sampling

```

```

;      CDQ2001 & CDQ2012 @ 48 and 32 K sampling
;      CDQ2012 & CDQ1000 @ 24 K sampling as per old CCS CDQ's
;      1 = 8-bit data byte count:
;      frames coded @ 44.1 sampling rate
;      frames coded with the sampling rate id bit equal to 0
;      MPEG-ISO @ 24, 22.05 and 16 K sampling
;      reed solomon frames

    clr     a          #smplidbit,r0    ;to init type of ancillary data count
;      & set addr of sampling rate id bit
    move    a,y:anctype    ;init ancillary count type as old CCS
    jset    #0,y:(r0),_try_441    ;if not low sample rate id, try 44.1

;set the flag to output ancillary data byte count vs pad byte count

_set_data_cnt
    bset    #0,y:anctype    ;set to do data byte count
    jmp     <_set_psych_parms    ;continue: psychoacoustic parameters

_try_441
    move    y:smplrte,x0    ;to test for 44.1 sampling rate
    move    #>SAM44K,a      ;set 44.1 code
    cmp     x0,a          #reedsolomon,r0 ;see if sample rate is 44.1
;      & set up to try reed solomon
    jeq     <_set_data_cnt    ;if 44.1, set data byte count type
    jset    #0,y:(r0),_set_data_cnt ;if reed solomon frames, data byte cnt

_set_psych_parms
;used on the sampling rate from XCODE:
;set the pyscho acoustic table of parameters

    move    #psychtable,r0    ;addr of psycho acoustic parameters
    move    #PSYCHBYSAMPLE,n0 ;num parameters per sample rate
    move    y:smplrte,b      ;to see if need to adjust address
    tst     b          #ptable,r1 ;if code 0, no need to shift address
;      & set address of operational table
    jeq     <_ptable_copy    ;if 0, get the table parameters

;adjust the table address to proper sampling rate psycho acoustic parameters

    rep     b
    move    (r0)+,n0

_ptable_copy
;save address of current sample rate psychtable for host vector update

    move    r0,y:psychaddr

;for the number of parameters copy sampling rate values to working table

    do      #PSYCHBYSAMPLE,_ptable_full
    move    y:(r0)+,x0
    move    x0,y:(r1)+

_ptable_full
;calculate buffer length controls
;for reed solomon set up for a three frame buffer for scale factor srs-12's

```

```

move    #reedsolomon,r4      ;to see if reed solomon applies
move    y:<outmus,y1          ;get the words per frame
move    #>2,x1               ;standard is a 2 frame output buffer
jclr    #0,y:(r4),_not_reed_a ;if not reed solomon, 2 frame buffer
move    #>3,x1               ;for reed solomon make a 3 frame buffer

_not_reed_a
mpy      x1,y1,a #>1,x1      ;set the mod buffer for numb frames
asr      a                   ;align integer result
move     a0,a                ;shift integer result
sub      x1,a                ;(frame numb words * numb) - 1

;now save the above buffer control values

move     a1,y:<outsize        ;set circular buffer ctl for o/p buffer

;set the type of frame as determined by the switches above

move     #>BOUND_4,x0        ;stereo intensity default to 4
move     x0,y:<sibound        ;save for frame header info
move     #>INTENSITY_4,x0    ;stereo intensity code for default of 4
move     x0,y:stintns        ;save for frame header info

;;;determine the type of framing STEREO vs MONO
;;
;; move     y:z3_psych,y1      ;init with MONO band-width
;; jset     #STEREO_vs_MONO,y:<stereo,_star_10 ;if mono, continue
;; move     y:s_psych,y1      ;else, get FULL stereo band-width
;; jclr     #JOINT_FRAMING,y:<stereo,_star_10 ;if joint bit allocation,
;; move     y:z4_psych,y1      ;else, get JOINT stereo band-width
;;
;;_star_10
;; move     y1,y:<usedsb       ;set used sub-band width

;calculate the b_i, ThresSLB & Thres10SLB tables for the selected sampling rate
; build b_ii table

move     #0,x0               ;b_i starting value
move     y:xaxisincr,x1      ;sample rate X-axis increment value
move     #BarkX,r0           ;address of array of b_i X values
move     #BarkY,r1           ;address of array of b_i Y values
move     #BarkS,r2           ;address of array of b_i YP values
move     #BrSScl,r5          ;address of b_i YP scale factor
move     y:NBark,r6          ;number of points in look up table
move     #>512,a             ;number of values to develop
move     #b_ii,r3            ;address of b_ii table to be built
jsr      mkBark              ;build the b_i table

; build ThresSLB table

move     #0,x0               ;ThresSLB starting value
move     y:xaxisincr,x1      ;sample rate X-axis increment value
move     #0.0/192.66,y0      ;threshold adjustment in slb
move     #ThresX,r0          ;address of array of ThresSLB X values
move     #ThresY,r1          ;address of array of ThresSLB Y values
move     #ThresS,r2          ;address of array of ThresSLB YP values
move     #ThSScl,r5          ;address of ThresSLB YP scale factor
move     y:NThres,r6         ;number of points in look up table
move     #>512,a             ;number of values to develop

```

```

        move    #ThresSLB,r3          ;address of ThresSLB table to be built
        jsr     mkthslb              ;build the ThresSLB table

; build Thres10SLB table (10 dB down table)

        move    #0,x0                ;ThresSLB starting value
        move    y:xaxisincr,x1       ;sample rate X-axis increment value
        move    #-12.04/192.66,y0     ;threshold adjustment in slb 10 dB down
        move    #ThresX,r0           ;address of array of ThresSLB X values
        move    #ThresY,r1           ;address of array of ThresSLB Y values
        move    #ThresS,r2           ;address of array of ThresSLB YP values
        move    #ThSScl,r5           ;address of ThresSLB YP scale factor
        move    y:NThres,r6          ;number of points in look up table
        move    #>512,a              ;number of values to develop
        move    #Thres10SLB,r3       ;address of Thres10SLB table to be built
        jsr     mkthslb              ;build the Thres10SLB table

; set the proper XPSYCHO table addresses based on sampling rate

        move    y:smplrte,a          ;get the sampling rate code
        move    #>SAM48K,x0          ;to test for 48 K sampling
        cmp     x0,a    #>SAM44K,x0  ;test for 48 K
        jeq     <_samp_48            ; & set up to test for 44.1 K sampling
        cmp     x0,a    #>SAM32K,x0  ;set up for 48 K sampling
        jeq     <_samp_44            ;test for 44.1 K
        cmp     x0,a    #>SAM24K,x0  ; & set up to test for 32 K sampling
        jeq     <_samp_32            ;set up for 44.1 K sampling
        cmp     x0,a    #>SAM22K,x0  ;test for 32 K
        jeq     <_samp_32            ; & set up to test for 24 K sampling
        cmp     x0,a    #>SAM22K,x0  ;set up for 32 K sampling
        jeq     <_samp_24            ;test for 24 K
        cmp     x0,a    #>SAM16K,x0  ; & set up to test for 22.05 K sampling
        jeq     <_samp_22            ;set up for 24 K sampling
        cmp     x0,a    #>SAM16K,x0  ;test for 22.05 K
        jeq     <_samp_22            ; & set up to test for 16 K sampling
        cmp     x0,a    #>SAM16K,x0  ;set up for 22.05 K sampling

_samp_16
; set up for 16 K Sampling

        move    #cb_16k,r4           ;address of cb table @ 16 K sampling
        move    #g_cb_16k,r5         ;address of g_cb table @ 16 K sampling
        jmp     <_cont_00

_samp_22
; set up for 22.05 K Sampling

        move    #cb_22k,r4           ;address of cb table @ 22.05 K sampling
        move    #g_cb_22k,r5         ;addr of g_cb table @ 22.05 K sampling
        jmp     <_cont_00

_samp_24
; set up for 24 K Sampling

        move    #cb_24k,r4           ;address of cb table @ 24 K sampling
        move    #g_cb_24k,r5         ;address of g_cb table @ 24 K sampling
        jmp     <_cont_00

```

\_samp\_32

; set up for 32 K Sampling

```

move    #cb_32k,r4          ;address of cb table @ 32 K sampling
move    #g_cb_32k,r5        ;address of g_cb table @ 32 K sampling
jmp     <_cont_00

```

\_samp\_44

; set up for 44.1 K Sampling

```

move    #cb_44k,r4          ;address of cb table @ 44.1 K sampling
move    #g_cb_44k,r5        ;address of g_cb table @ 44.1 K sampling
jmp     <_cont_00

```

\_samp\_48

; set up for 48 K Sampling

```

move    #cb_48k,r4          ;address of cb table @ 48 K sampling
move    #g_cb_48k,r5        ;address of g_cb table @ 48 K sampling

```

\_cont\_00

; set the standard threshold of hearing tables

```

move    #ThresSLB,r0

```

; set threshold of hearing table address as per switches bitrate &amp; frame type

```

move    y:thresh,a          ;get flag based on switches
tst     a                   ;test the flag
jne     <_cont_10           ;if non-zero, go with standard tables

```

; use the threshold of hearing table that is 10 db down

```

move    #Thres10SLB,r0

```

\_cont\_10

;set the sampling rate table addresses for this frame

```

move    r0,y:thresslb       ;set active selected table address
move    r0,y:holdthresslb   ;save the selected table address
move    #ThresSLB,x0
move    x0,y:splitthresslb  ;set split mono table address
move    #b_i,r2             ;address of b_i table
move    r2,y:b_i
move    #fmap_x,r3          ;address of fmap table
move    r3,y:fmap
move    r4,y:cb
move    r5,y:g_cb

```

```

; set output write read pointer to something safe since interrupts will
; be on before it is set properly.

```

```

move    #framebuf,r0        ;address of the output frame buffer
move    r0,y:<oprptr        ;set the output read buffer

```

144

BAD ORIGINAL



```
; set up for ancillary data to be decoded from a framed and transmit via rs232
; a. zero the input data byte counter and bytes for current frame
; b. set address of clock table, baudclk, based on baud rate (0 thru 7)
; c. set table offset by baud rate;
; these are standard CDQ2000 set by macro, BAUDCLK, in box_ctl.asm):
; 0 = 300 baud
; 1 = 1200 baud
; 2 = 2400 baud
; 3 = 3200 baud
; 4 = 4800 baud
; 5 = 38400 baud
; 6 = 9600 baud
; 7 = 19200 baud
; d. set transmit enable (for xon/xoff)
; e. get and set the clock for baud rate from the table
; f. get and set the max bytes for baud rate from the table
; g. set the data input and output pointers
; h. set receive enable
; i. set receive enable interrupt
```

```
move    #0,x0                ;zero the received data counter
move    x0,y:bytecnt          ;zero the byte counter
move    x0,y:bytesfrm         ;zero the current frame byte counter
move    #bauddata,r0          ;get data baud rate table address
move    #DATABYBAUDRATE,n0     ;number parameters per baud rate
move    y:baudrte,b           ;to see if need to adjust address
tst     b                     ;if code 0, no need to shift address
jeq     <_baudrte             ;if 0, get the clock parameter
```

```
; adjust the table address to proper ancillary data baud rate parameters
```

```
rep     b
move    (r0)+n0
```

```
_baudrte
```

```
move    y:(r0)+,r2            ;get clock value at baud rate
; & move to code 0 max bytes per frame
move    y:smplrte,n0          ;sample rate is now offset to max bytes
nop
move    y:(r0+n0),x0           ;frame max bytes for check of bytecnt
move    x0,y:maxbytes          ;store maxbytes for scixmt to check
move    #databytes,x0          ;get addr of the data byte buffer
move    x0,y:dataiptr          ;address for next byte received
move    x0,y:dataoptr          ;addr for next byte to output to frame
movep    r2,x:<<M_SCCR          ;set the clock for selected baud rate
bset    #M_RE,x:<<M_SCR         ;set receive enable
bset    #M_RIE,x:<<M_SCR        ;data expected set receive interrupt
bset    #M_TE,x:<<M_SCR         ;set transmit enable
```

```
; enable the host command interrupt
```

```
bset    #M_HCIE,x:<<M_HCR
```

```
; Set and clear a flag so we can set the scope trigger.
```

```
ON BITALLOC_LED_CD            ;set a different flag for debug
OFF_BITALLOC_LED_CD
```

```
; Now form the two pointers to the output buffer.
```

145

BAD ORIGINAL

; y:<frmstrt - is the write pointer for the 1st frame to be coded.  
 ; y:<oprptr - is the read pointer to start with the 2nd frame.  
 ; frmstrt is used to point to where the current buffer is for outputting  
 ; data into. This data is a result of the current musicam coding.

```

move    #framebuf,r0      ;address of the output frame buffer
move    y:<outmus,n0       ;set the output read ptr
move    y:<outsize,m0      ;set the output buffer circular ctl
move    r0,y:<frmstrt      ;1st frame at start of buffer
move    (r0)+n0           ;advance to start of 2nd frame
move    r0,y:<frmnxt       ;set to align with timer (frame sync)
move    r0,y:<oprptr       ;set the output read buffer
move    #$ffff,m0        ;reset to linear buffer

```

;start the padded frame REST variable at zero

```

clr     a
move    a,y:padrest

```

;start the output framing for setvalue at beginning of the frame buffer

```

jsr     bitsallo

```

;clear the frame sync received flag and zero the time out counter

```

bclr    #0,y:<timer        ;clear frame sync flag
clr     a
move    a,y:<timeout       ;zero frame sync timed failure counter

```

```

; IRQA set to IPL 3, negative edge (lowest priority)
; JSI set to IPL 3
; IRQB set to IPL 3, negative edge (highest priority)
; SCI set to IPL 3
; HOST set to IPL 2
; all same priority

```

```

; movep    #>$f03f,x:<<M_IPR      ;set int priorities and edges
; movep    #>$f83f,x:<<M_IPR      ;set int priorities and edges

```

;wait for the dust to settle before pushing onward

```

move    #>XCODE_STARTUP,a
jsr     wait

```

```

SET_ADC_RESET      ;stop A to D calibration

```

```

movep    x:<<M_SR,a          ;clear the exception
movep    #0,x:<<M_TX        ;output the data

```

```

andi     #$fc,mr          ;turn on the interrupt system

```

```

;stitle   'Main Code'
;page

```

;main loop capturing frames of audio, performing the psychoacoustic analysis  
 and encoding MUSICAM frames of audio data that are sent to a MUSICAM decoder

top

```

;!!!dbg bset    WATCH_DOG      ;tickle the dog

```

146

BAD ORIGINAL



```

;!!!dbg bclr    WATCH_DOG                ;tickle the dog
      bset     #1,x:<<SFFES              ;!!!dbg: WATCH_DOG
      bclr     #1,x:<<SFFES              ;!!!dbg: WATCH_DOG

      ;check for time out since last frame sync pulse

      move     y:<timeout,a                ;get top loop counter
      move     #>TOP_TIMEOUT_CD,x0        ;get failure value
      cmp      x0,a    #>1,x0              ;test counter versus failure value
      ; & set up to increment counter
      jlt      <_top_1                    ;if not failed, continue
      bclr     #1,y:<timer                ;clear the 1st frame skipped flag
      jmp      restart                    ;restart and skip the 1st frame

_top_1
;increment the failure counter

      add      x0,a                        ;increment counter
      move     a,y:<timeout                ;save counter so far this loop

;get the external switches to determine if any changes that signal a restart

      GET_SWITCHES_CD gsws_10
      jsr      getsws
      jclr     #4,y:<not_appl,_lets_go

;!!!dbg
;      jmp      restart                    ;!!! debug
;!!!dbg

;we have to restart with new framing criteria,
;protect the decoding of frames by clearing 2 successive frame

      move     y:<frmstrt,r6                ;set starting for output buffer
      move     y:<outsize,m6                ;set the output buffer circular ctl
      clr      a    #reedsolomon,r4        ;to zero the output frames buffers
      ; & to see if reed solomon applies

      do       y:<outmus,_clear_1            ;clear the 1st frame
      move     a,y:(r6)+

_clear_1
      jclr     #0,y:<timer,_clear_1          ;check for new frame
      bclr     #0,y:<timer
      do       y:<outmus,_clear_2            ;clear the 2nd frame
      move     a,y:(r6)+

_clear_2
      jclr     #0,y:<timer,_clear_2          ;check for new frame
      bclr     #0,y:<timer
      jclr     #0,y:(r4),_clear_done        ;if not reed solomon, 2 frame buffer
      do       y:<outmus,_clear_3            ;clear the 3rd frame
      move     a,y:(r6)+

_clear_3
      jclr     #0,y:<timer,_clear_3          ;check for new frame
      bclr     #0,y:<timer

_clear_done
      move     #-1,m6                        ;restore r6 to linear buffer control
      jmp      restart                    ;let's start anew

_lets_go

;test to light AES-EBU led

```

147

BAD ORIGINAL

## XPSYCHO\_AES\_EBU\_TEST

; initialize stereo control settings to reflect current transmission

```
jsr      setctis

jclr     #0,y:<timer,top      ;check for new frame
bclr     #0,y:<timer
bclr     #0,y:<qttalloc        ;clear frame msec timer bit allocation

bclr     #1,x:<<$FFES        ;!!!dbg: WATCH_DOG
```

; zero the time out counter  
; and see if this is the first frame since last restart,  
; and if so, set 1st frame flag and restart again

```
clr      a
move     a,y:<timeout          ;reset the time out counter
jset     #1,y:<timer,_lets_go_2 ;if 1st frame bypassed, continue
bset     #1,y:<timer          ;indicate skipped the 1st frame
jmp      restart
```

\_lets\_go\_2

; toggle the host watch dog flag

## TOGGLE\_WATCH\_DOG\_CL

```
!!!dbg: debug the encoding of frame when a frame count limit is reached
move     y:dbgcnt,a           ;!!!dbg: get debug frame count
move     #>1,x0               ;!!!dbg: to increment debug frame count
add      x0,a      #>40,x0    ;!!!dbg: increment debug frame counter
;                                     ;!!!dbg: & get count limit to start debugging
cmp      x0,a                 ;!!!dbg: see if frame count reached limit
jlt      <_dbg_cnt_0          ;!!!dbg: if not at limit, save new count
!!!dbg: debug limit reached: turn off interrupts and encode what we've got now
ori      #S03,mr
nop
nop
nop
clr      a                   ;!!!dbg: zero the frame counter
_dbg_cnt_0
move     a,y:dbgcnt           ;!!!dbg: save new debug frame count value
!!!dbg
```

; set the working value for padding calculation

```
move     y:paddiff,x0         ;get normal DIFF value for pad calc
move     x0,y:usediff         ;init as pad calc diff value
```

; set the joint boundary determination controls:

```
minimum joint sub-band to set boundary
left and right channel anti correlation tolerance value
minimum sub-band requiring at least 1 index allocation
```

```
move     y:maxsubbands,x0     ;get normal MAXSUBBANDS
move     x0,y:<maxsubs         ;set the working MAXSUBBANDS
move     y:z3_psych,a         ;init with MONO band-width
jset     #STEREO_vs_MONO,y:<stereo,_xxxx_10 ;if mono, continue
```

148

BAD ORIGINAL

```

        move     y:s_psych,a           ;else, get FULL stereo band-width
        jclr     #JOINT_FRAMING,y:<stereo,_xxxx_10 ;if joint bit allocation,
        move     y:z4_psych,a         ;else, get JOINT stereo band-width

;xxx_10
        tst      a                     ;see if used sub-bands would be zero
        jeq      <_use_maxsubs        ;if zero, reset to maxsubbands
        cmp      x0,a a,y:<usedsb      ;see if table sub-bands ok vs maxsubs
        ; & in case, set usedsb to table value
        jle      <_aft_10             ;table value ok, continue

;_use_maxsubs
;default used sub-band width to maxsubs
        move     x0,y:<usedsb

;_aft_10
;calculate the b_i, ThresSLB & Thres10SLB tables for the selected sampling rate

        move     y:k_psych,x0          ;get minimum joint sub-band
        move     x0,y:jntmin           ;set minimum for next frame
        move     y:l_psych,x0          ;get 2 channels anti correlation value
        move     x0,y:jntanti          ;set anti correlation tolerance value
        move     y:m_psych,x0          ;get minimum sub-band req 1 allocation
        move     x0,y:<limitsb          ;set LIMITSUBBANDS for next frame
        move     y:t_psych,x0          ;get joint frame decrement count value
        move     x0,y:jntfrms          ;set joint frame decrement count value

;get the selected DbAddTbl (@ 3db or 6db)

        move     y:v_psych,a           ;get the selection variable
        move     #.5,x0                ;get the test value
        cmp      x0,a #DbAddTbl_3db,x0 ;less than half is 3db table
        ; & in case, set DbAdd table @ 3db
        jlt      <_cont_11            ;if less, set the working table address

;the DbAdd table @ 6 db was selected
        move     #DbAddTbl_6db,x0

;_cont_11
;set working table address for DbAddTbl
        move     x0,y:dbaddtbl

;if doing a split mode of transmission:
;
;    set framing controls
;    set the frame header bit rate code
;    set the frame size in words and bits
;    set the applicable bit allocation control parameters

        jclr     #SPLIT_MODE,y:<stereo,_top_60

        move     y:frmttype,a           ;get specified framing via switches
        move     y:maxsubbands,b        ;get normal MAXSUBBANDS
        move     y:holdthressib,y0      ;selected ThresSLB table addr
        move     y:holdthreshld,y1      ;selected Threshld table addr

```

```

; if we are doing a split mono frame, set the output frame type to mono

        jclr      #SPLIT_MONO_FRAME,y:<stereo,_top_05 ;if not appl, continue
        move      #>MONO,a
        move      y:spltmaxsubs,b                ;get split rate MAXSUBBANDS
        move      y:splitthresslb,y0             ;split mono ThresSLB table addr
        move      y:splitthreshld,y1             ;split mono Threshld table addr
; ;
; ???         move      y:splt paddiff,x0          ;get split DIFF value for pad calc
; ???         move      x0,y:usediff              ;set DIFF value for pad calc

_top_05
        move      a,y:opfirtyp                    ;set current frame type for output to
                                                ; the coded frame (this can change
                                                ; from frame to frame from JOINT STEREO
                                                ; to FULL STEREO if the JOINT STEREO bit
                                                ; allocation applies and can handle the
                                                ; curr frame data as true full stereo)

        move      b,y:<maxsubs                    ;set working MAXSUBBANDS
        move      y0,y:thresslb                  ;set active ThresSLB table addr
; ;
        move      y1,y:threshld                  ;set active Threshld table addr

; initialize proper control flags:

        bclr      #STEREO_vs_MONO,y:<stereo ;default to stereo
        bclr      #JOINT_FRAMING,y:<stereo ;default to NOT joint stere

        move      #>MONO,x0                      ;start with mono
        cmp       x0,a      #>JOINT_STEREO,x0 ;compare and set up for JOINT
        jne       <_top_10

; indicate mono framing (default is stereo)

        bset      #STEREO_vs_MONO,y:<stereo

        jmp       <_top_20

_top_10
        cmp       x0,a                          ;if not JOINT, defaults to full stereo
        jne       <_top_20

; indicate joint stereo framing (default is not joint)

        bset      #JOINT_FRAMING,y:<stereo

_top_20

; determine the sub-band ranges
; if applicable, setup the 128 or 112 Kbits split frame mono

        jset      #SPLIT_MONO_FRAME,y:<stereo,_top_30

; otherwise, normal 128 or 112 frame

        move      y:s_psych,n0                   ;num sub-bands for FULL STEREO
        move      y:z4_psych,n1                  ;num sub-bands for JOINT STEREO
        move      y:z3_psych,n2                  ;num sub-bands for MONO
        jmp       <_top_40

_top_30

```

150

BAD ORIGINAL

```
;set the used sub-bands based on the split mono bit rate
```

```
    move    #11,n2          ;!!!dbg: for now force 11 usedsubbands
```

```
_top_40
```

```
;determine the type of framing STEREO vs MONO
```

```
    move    n2,a            ;init with MONO band-width
    jset    #STEREO_vs_MONO,y:<stereo,_top_50 ;if stereo bit allocation,
    move    n0,a            ; get FULL stereo band-width
    jclr    #JOINT_FRAMING,y:<stereo,_top_50 ;if joint bit allocation,
    move    n1,a            ; get JOINT stereo band-width
```

```
_top_50
```

```
    tst     a               y:<maxsubs,x0    ;see if used sub-bands would be zero
                                           ; & get maxsubbands to test
    jeq     <_use_maxsubs_a    ;if zero, reset to maxsubbands
    cmp     x0,a             a,y:<usedsb     ;see if table sub-bands ok vs maxsubs
                                           ; & in case, set usedsb to table value
    jle     <_aft_10_a        ;table value ok, continue
```

```
_use_maxsubs_a
```

```
;default used sub-band width to maxsubs
```

```
    move    x0,y:<usedsb
```

```
_ft_10_a
```

```
_top_60
```

```
;start of XPSYCHO processing:
```

```
; start with the left channel:
```

```
    bclr    #LEFT_vs_RIGHT,y:<stereo
```

```
_chan_2nd_
```

```
;come back here to analyze the 2nd channel
```

```
; Now get the position to read the fft data from
```

```
; This buffer is offset from the polyphase filter to account for the
; delay through the filter.
```

```
    move    #PCMSIZE*2-1,m0    ;set as a mod buffer for both channels
    move    x:<polyst,r0       ;get input pcm buffer address
```

```
;test for need to adjust for 2nd channel
```

```
    jclr    #LEFT_vs_RIGHT,y:<stereo,_hann_00
    move    (r0)+              ;advance to 2nd channel
```

```
_hann_00
```

```
    move    #(256-64),n0      ;back up to position fft
    move    #hbuf,r1          ;get hanning output buffer address
    move    (r0)-n0           ;back-up one channel
    move    (r0)-n0           ;back-up another channel
```

```

move    #2,n0                ;set offset for two channels
jsr     hanning              ;apply a hanning window
move    #-1,m0               ;restore r0 to linear buffer
jsr     fft                  ;fft the data
move    #fftbuf,r0           ;real part of fft
move    #fftbuf,r4           ;imaginary part of fft
move    #power,r1            ;power array
move    #hbuf,r0             ;compute power of fft data
jsr     logpow

move    #power,r0            ;power array
move    #SBMaxDb,r1          ;maximum in each sub-band (slb)
move    #NUMSUBBANDS,n1      ;in case it's the 2nd channel

;test for left or right channel currently to set address for sub-band max values
jclr    #LEFT_vs_RIGHT.y:<stereo,_sbmax_00

;adjust address for 2nd channel
move    (r1)+n1              ;offset to 2nd channel part of array

_sbmax_00
jsr     findmaxi             ;find max power in a sub-band
move    #power,r1            ;power array
move    #Tonals,r2           ;tonal array
move    #rngtbl,r4           ;range table for tonal search
jsr     findtona             ;find tonals
move    r3,x:<ntonals        ;save number of tonals

move    #power,r1            ;power array
move    #Tonals,r2           ;tonal array
move    #rngtbl,r4           ;range table for tonal search
jsr     zeropowe             ;zero power around tonals

move    #power,r1            ;power array
move    #NoisePwr,r2         ;address of the noise array
jsr     findnois            ;find the noise

move    #Maskers,r3          ;address of the masker structure
move    #NoisePwr,r2         ;address of the noise array
move    #Tonals,r1           ;address of the Tonals structure
move    x:<ntonals,x0        ;# of tonals in Tonals structure
jsr     mergemas            ;merge the maskers
move    b,x:<nmasker         ;save # of maskers

move    #Maskers,r0          ;address of the masker structure
move    x:<nmasker,b         ;number of maskers in masker structure
jsr     finddbma            ;find the db value of maskers

move    #Maskers,r0          ;address of the masker structure
jsr     pruneclo            ;prune close maskers

move    #Maskers,r0          ;address of the masker structure
move    x:<nmasker,b         ;number of maskers in masker structure

```

```

        jsr      prunequi          ;prune quiet maskers

        move     #Maskers,r0        ;address of the masker structure
        move     x:<nmasker,b       ;number of maskers in masker structure
        jsr      prunemas          ;prune masked maskers

        jsr      findalis          ;find alising components

;if doing left or right channel currently to save number aliasers accordingly
        jset     #LEFT_vs_RIGHT,y:<stereo,_alis_00

        move     b,x:nalislft      ;save left channel in local memory
        jmp      <_alis_10

_alis_00
        move     b,x:nalisrgt      ;save right channel in local memory

_alis_10
        move     #Maskers,r4        ;address of the masker structure
        move     #GlbMsk,r1        ;address of global masking threshold
        move     #MAXNMSKFREQS,n1   ;in case it's the 2nd channel

;if left or right channel currently to set address for sub-band global mask
        jclr     #LEFT_vs_RIGHT,y:<stereo,_gbmsk_00

;adjust address for 2nd channel
        move     (r1)+n1           ;offset to 2nd channel part of array

_gbmsk_00
        jsr      QCalcGlo          ;calculate global masking threshold

; test Maskers array for a sine wave
        move     #Maskers,r0        ;address of the masker structure
        jsr      tststine          ;check maskers for a sine wave

        move     x:<sincnt,y0       ;get the updated sine frame count
        move     x:<sintest,y1      ;get the updated sine flag

;test if doing left or right channel currently to save result of sine wave test
        jset     #LEFT_vs_RIGHT,y:<stereo,_sine_00

        move     x0,y:strtsinlft   ;save left channel in local memory
        move     x1,y:endsinlft    ;save left channel in local memory
        move     y0,y:sincntlft    ;save left channel frame counter
        move     y1,y:sintstlft    ;save left channel sine flag
        jmp      <_sine_10

_sine_00
        move     x0,y:strtsinrgt   ;save right channel in local memory
        move     x1,y:endsinrgt    ;save right channel in local memory
        move     a,y:sincntrgt     ;save right channel frame counter
        move     y1,y:sintstrgt     ;save right channel sine flag

_sine_10

```

153

BAD ORIGINAL

```

; polyphase filter the input data

    move    x:<polyst,r0          ;get polyana start address-left channel
    move    #2,n0                ;set offset for two channels
    move    #PCMSIZE*2-1,m0      ;set as a mod buffer for both channels
    move    y:curxlft,r3         ;left channel current x vector location
    move    #polydta,r5          ;left chan output buffer poly analyzed
    move    #INPCM,n5            ;in case of right channel

;test for left or right channel currently to set address for poly analyzed data

    jclr    #LEFT_vs_RIGHT,y:<stereo,_panal_00

    move    (r0)+                ;advance start inpcm data to right chan
    move    y:curxrgt,r3         ;right channel curr x vector location

;adjust address for right channel

    move    (r5)+n5              ;offset right channel poly analyzed data

_panal_00
    jsr     polyanal             ;poly analyze the data
    move    #-1,m0               ;restore to a linear buffer

;if left or right channel currently to save address in x vector accordingly

    jset    #LEFT_vs_RIGHT,y:<stereo,_panal_20

;save left channel current position in x vector buffer

    move    r3,y:curxlft         ;left channel current x vector location
    jmp     <_panal_30

_panal_20
;save right channel current position in x vector buffer

    move    r3,y:curxrgt         ;right channel current x vector location

_panal_30

; develop the scale factors
; initialize the table of scale factors to minimum amplitude (-63 ==> 0 ampl)

    move    #SBndSKF,r1          ;left chan addr of subband scale factors
    move    #0,n0                ;start with left channel polydata
    move    #0,n1                ;start with left channel scale factors

;test for left or right channel currently to set address for scale factors

    jclr    #LEFT_vs_RIGHT,y:<stereo,_scfct_00

;adjust address for right channel

    move    #NUMSUBBANDS*NPERGROUP,n1 ;right channel scale factors
    move    #INPCM,n0             ;right channel poly analyzed data
    move    (r1)+n1              ;offset to right channel part of array

_scfct_00

```

154

BAD ORIGINAL



; initialize the scale factor array with the terminal value of 63

```

move    #63,n4
do      #NUMSUBBANDS*NPERGROUP,_init_00
move    n4,x:(r1)+      ;store 63 in scale factor array

```

\_init\_00

```

move    #polydta,r0      ;addr of poly analyzed data
move    #SBndSKF,r1      ;addr of sub-band scale factors
move    (r0)+n0          ;offset to proper channel poly data
move    (r1)+n1          ;offset to proper channel scale factors
jsr     findskf          ;find scale factors

```

; develop the SBits for scale factors

```

move    #SBndSKF,r0      ;left chan addr sub-band scale factors
move    #SBits,r1        ;left channel addr of sub-band sbits

```

; test for left or right channel to set addresses for scale factors and sbits

```

jclr    #LEFT_vs_RIGHT,y:<stereo,_sbits_00

```

; adjust address for right channel

```

move    #NUMSUBBANDS*NPERGROUP,n0 ;right channel scale factors
move    #NUMSUBBANDS,n1          ;right channel scale factor sbits
move    (r0)+n0                  ;offset to right channel part of array
move    (r1)+n1                  ;offset to right channel part of array

```

\_bits\_00

```

jsr     pickskf          ;pick the best scale factors

```

; set correct maximum level for the channel

```

move    #SBndSKF,r0      ;left chan addr sub-band scale factors
move    #SBMaxDb,r1      ;left maximum in each sub-band (slb)
move    #polydta,r2      ;left chan poly analyzed buffer

```

; test for left or right channel to set addresses for scale factors and SBMaxDb

```

jclr    #LEFT_vs_RIGHT,y:<stereo,_cksub_00

```

; adjust address for right channel

```

move    #NUMSUBBANDS*NPERGROUP,n0 ;right channel scale factors
move    #NUMSUBBANDS,n1          ;right channel scale factor sbits
move    #INPCM,n2              ;right channel poly analyzed data
move    (r0)+n0                ;offset to right channel part of array
move    (r1)+n1                ;offset to right channel part of array
move    (r2)+n2                ;offset to right channel part of array

```

\_cksub\_00

; determine which method to use to determine the sub-band maximum values

```

move    y:u_psych,a      ;get use findrms.asm rtn parameter
move    #.5,x1            ;if less than .5, use checksub.asm rtn
cmp     x1,a              ;see if parameter less than .5
jlt     <_do_checksub     ;if less, use checksub.asm rtn

```



;use RMS for maximum level for the sub-band

```

move    r2,r0                ;addr of poly analyzed data
jsr     findrms              ;find max in a subband
jmp     <_set_min_mask       ;go to set minimum masking level

```

\_do\_checksusb

; set correct maximum level for the channel

```

jsr     checksusb           ;find max in a subband

```

\_set\_min\_mask

; set minimum masking level in each sub-band

```

move    #MinMskDb,r1        ;minimum masking per subband (slb)
move    #NUMSUBBANDS,n1     ;right channel minimum masking per subs

```

;test for left or right channel to set address

```

jclr    #LEFT_vs_RIGHT,y:<stereo,_ckmin_00

```

;adjust address for right channel

```

move    (r1)+n1             ;offset to right channel part of array

```

\_ckmin\_00

```

move    #GlbMsk,r0          ;global masking threshold
jsr     findminm            ;find min masking

```

; set minimum masking level in each sub-band

```

move    x:nalislft,a        ;left channel number of aliasers
move    #Alising,r0         ;left channel aliasing structure
move    #SBMaxDb,r1         ;left channel max in each sub-band (slb)

```

;test for left or right channel to set addresses

```

jclr    #LEFT_vs_RIGHT,y:<stereo,_ckmax_00

```

```

move    x:nalisrgt,a        ;right channel number of aliasers

```

;adjust address for right channel

```

move    #MAXTONALS*ALIASSIZE,n0 ;right channel alias structures
move    #NUMSUBBANDS,n1        ;right channel sub-band max's
move    (r0)+n0                ;offset to right channel part of array
move    (r1)+n1                ;offset to right channel part of array

```

\_ckmax\_00

```

jsr     findmaxs            ;find the maximum signal

```

we're doing mono frames,  
skip the right channel XPSYCHO processing

```

jset    #STEREO_vs_MONO,y:<stereo,_xcode_00 ; if doing mono, skip right

```

```

; if we've done the left channel, initialize and go back for the right channel
    jset    #LEFT_vs_RIGHT,y:<stereo,_xcode_00 ; if did right, continue
; flag for right channel
    bset    #LEFT_vs_RIGHT,y:<stereo ; indicate right channel
    jmp     <_chan_2nd_ ; go back & do same stuff for right chan
_xcode_00
; We have now finished all XPSYCHO the processing.
; set the working frame length in bits and handle any required padded frame
; determination
    jsr     setframelen
; set number of fixed bits required, and the number of available bits for audio
; just in case we are doing JOINT framing, set the flag to determine the
; fixed and available bits for full stereo
    bset    #JOINT_at_FULL,y:<stereo ; to develop FULL bits available
    jsr     bitpool
    bclr    #JOINT_at_FULL,y:<stereo ; clear after setting FULL bits
    move    x0,y:<fixBits ; save fixed bit count
    move    x1,y:<audbits ; save bit count available for alloc
; !!!dbg
;     nop
;     nop
;     nop
;     nop
;     nop
;     jmp     top ; !!! debug if using stored frames buffer
; !!!dbg
; test if a sine wave in either or both channels according to XPSYCHO's
; initialize as NOT a sine wave in either channel
    bclr    #LEFT_SINE_WAVE,y:<stereo
    bclr    #RIGHT_SINE_WAVE,y:<stereo
; if the starting sub-band == -1, NOT a sine wave
    move    #>-1,x0 ; indicator NOT a sign wave
    move    y:strtsinlft,a ; left sine wave start subband
    cmp     x0,a y:endsinlft,y0 ; to see if not a sine wave
    ; & save Left EndSin
    jeq     <_sin_R1 ; if not a sine wave, try right channel
; set left channel sine wave flag and light the indicating LED
    bset    #LEFT_SINE_WAVE,y:<stereo
_sin_R1
    jset    #STEREO_vs_MONO,y:<stereo,_sin_R2 ; 1 chan. skip right channel
; if the starting sub-band == -1, NOT a sine wave

```

157

BAD ORIGINAL



```

        move    y:strtsinrgt,b           ;right sine wave start subband
        cmp     x0,b    y:endsinrgt,y1  ;to see if not a sine wave
                                           ; & save right EndSin
        jeq     <_sin_R2                ;if not a sine wave, do bit allocation

;set right channel sine wave flag and light the indicating LED
        bset    #RIGHT_SINE_WAVE,y:<stereo

_sin_R2
;:if both channels have a sine wave detected,
;:the audio input cannot be a test sine wave and must
;:be reset as true audio

        jclr    #LEFT_SINE_WAVE,y:<stereo,_sin_OK      ;NOT a sine wave, OK
        jclr    #RIGHT_SINE_WAVE,y:<stereo,_sin_OK     ;NOT a sine wave, OK

_kill_sin
;:since both channels have a sine wave,
;:re-initialize as NOT a sine wave in either channel

        bclr    #LEFT_SINE_WAVE,y:<stereo
        bclr    #RIGHT_SINE_WAVE,y:<stereo
        move    #>-1,x0                    ;indicator NOT a sign wave
        move    x0,y:strtsinlft           ;set as not a sine wave
        move    x0,y:endsinlft           ;set as not a sine wave
        move    x0,y:strtsinrgt          ;set as not a sine wave
        move    x0,y:endsinrgt           ;set as not a sine wave

_sin_OK
;:now see if there is a sine wave in one of the channels and if so,
;:see if the other channel has audio and if so, cancel the
;:sine wave in the one channel. The input does not qualify as
;:a test tone.

        jclr    #LEFT_SINE_WAVE,y:<stereo,_sin_RT

;left channel has a sine wave, get set to check for audio in right channel

        move    #SBndSKF,r0                ;to get scale factor min-left
        move    #SBndSKF,r1                ;to look for audio in right
        move    #NUMSUBBANDS*NPERGROUP,n1  ;offset to right
        move    y:strtsinlft,a
        move    y:endsinlft,b
        move    (r1)+n1                    ;to chk right channel for audio
        jclr    #RIGHT_SINE_WAVE,y:<stereo,_sin_lchan
        jmp     <_sin_OK2

_sin_RT
        jclr    #RIGHT_SINE_WAVE,y:<stereo,_sin_OK2

;right channel has a sine wave, get set to check for audio in left channel

        move    #SBndSKF,r0                ;set scale factor array
        move    #NUMSUBBANDS*NPERGROUP,n0  ;offset to right channel
        move    #SBndSKF,r1                ;to chk left channel for audio
        move    y:strtsinrgt,a

```

```

        move    y:endsinrgt,b
        move    (r0)+n0                ;to get scale factor min-right

_sin_1chan
;one channel has a sine wave, see if there is audio in the other channel

        tst     a        #NPERGROUP,n0    ;check for sub-band 0
                                           ; & by 3 scale factors per sb
        jeq     <_sin_min                ;if zero, go to get min skf

;advance to start sub-band of sine wave

        do      a,_sin_min
        move    (r0)+n0

_sin_min
;determine the number of sine wave scale facotrs to check for lowest

        sub     a,b        #>NPERGROUP,x0    ;calc number of sub-bands in range
                                           ; & to calculate entries per sub-band
        move    #>1,y0                ;to account for subtracted sub_band
        add     y0,b                ;incrment 1 sub-band
        move    b,y0                ;shift end sub-band to multiply reg
        mpy     x0,y0,a #NPERGROUP,n1    ;calculate number of scale factors
                                           ; & to skip sub-band 0 in other channel
        asr     a                ;align the integer result

        ;at the lowest scale factor over the sine wave range

        move    #>63,b                ;start with largest skf

;search over scale factor span from above

        do      a0,_get_min
        move    x:(r0)+,x0                ;get the scale factor
        cmp     x0,b                ;see if scale factor less than last
        tgt     x0,b                ;if so, set new lower scale factor

_get_min
;calculate the minimum scale factor value to look for in the other channel
;to see if audio is present

        move    #>SINE_SKF_TEST,x0        ;set the scale factor adjust to min
        add     x0,b                ;calc the minimum scale factor to test
        move    (r1)+n1

        do      #NUMSUBBANDS*NPERGROUP-3,_sin_OK2
        move    x:(r1)+,x0                ;get scale factor
        cmp     x0,b                ;test versus minimum
        jlt     <_cont_sin_tst        ;if not below minimum, not audio yet

;we have audion in the other channel, clear the sine wave indication

        enddo                        ;break the loop, it's audio
        jmp     <_kill_sin            ;clear the sine wave ind in one channe

_cont_sin_tst

```

nbp

\_sin\_OK2

we got here, we have a legitimate test tone (sine wave) in one channel  
and nothing in the other channel

;allocate the bits in the frame by subband

```

move    #SBits,r0           ;scale factors
move    #MirMskDb,r1        ;minimum masking per sub-band (slb)
move    #SBMaxDb,r2         ;maximum in each sub-band (slb)
move    #SBPos,r4           ;sub-band position
move    #SBindx,r5          ;sub-band indicies
jsr      bitalloc           ;allocate the bits

clr      a
move     a,y:<bitscnt        ;start the bit counter of framed bits
move     y:<frmstrt,r6       ;set starting for output buffer
move     y:<outsize,m6       ;set the output buffer circular ct!

jsr      setsync            ;set the sync bits

jsr      setsyst            ;set the system bits

```

;set framing mode led

```

move     y:opfrrtyp,a       ;get current frame's type via bitalloc

SET_FRAME_TYPE_LED_CD      ;light the proper leds

move     y:opfrrtyp,a       ;get current frame's type via bitalloc
move     #>MONO,x0          ;start with mono
cmp      x0,a #>JOINT_STEREO,x0 ;compare and set up for JOINT
jne      <_xcde_55

OFF_JOINT_LED_CD           ;clear the JOINT stereo led
OFF_STEREO_LED_CD         ;clear the FULL stereo led
ON_MONO_LED_CD             ;light the MONO led
jmp      <_xcde_57

```

```

_xcde_55
cmp      x0,a               ;if not JOINT, defaults to full stereo
jne      <_xcde_56

```

```

OFF_MONO_LED_CD            ;clear the MONO led
OFF_STEREO_LED_CD         ;clear the FULL stereo led
ON_JOINT_LED_CD            ;light the JOINT stereo led
jmp      <_xcde_57

```

```

_xcde_56
OFF_MONO_LED_CD            ;clear the MONO led
OFF_JOINT_LED_CD          ;clear the JOINT stereo led
ON_STEREO_LED_CD          ;light the FULL stereo led

```

```

de_57
SET_LEDS_CD

```

;if there is CRC-16 protection on the frame:  
; set the CRC-16 checksum bit count for the old ISO method:

160

BAD ORIGINAL

```

; a. header bits covered by any type of frame
;   plus bits for the left channel also apply to any type of frame
; b. set bits for possible right channel based on frame type
; c. if not MONO, add bits for right channel
; d. save old ISO bit count for this frame
; e. clear the space in the frame to zeroes

    jclr    #PROTECT,y:<stereo,_xcde_60      ;if no checksum, set allocations

    move    #>CRC_BITS_A+CRC_BITS_B,a
    move    #>CRC_BITS_B,x0                  ;bit count for right channels
    jset    #STEREO_vs_MONO,y:<stereo,_xcde_58
    add     x0,a                             ;since its stereo, add for right channel

_xcde_58
    move    a,x:crcold                      ;set the old ISO CRC-16 bit count
    jsr     clr crc                          ;clear the crc bits

_xcde_60
    move    #SBIndx,r0                      ;sub-band indicies
    jsr     setbal                           ;set the bit allocations

; if doing joint stereo,
;   from the intensity sub-band boundary thru the last used
;   sub-band, move the joint SBits, SKFs and the averaged poly samples
;   for setsbits, setskf and setdata routines
;   the SBits and SKfs are left and right channels
;   the poly samples are stored in the regular left channel samples

    jclr    #JOINT_FRAMING,y:<stereo,_xcde_699      ;Not Joint Framing
    jset    #JOINT_at_FULL,y:<stereo,_xcde_699      ;Joint upgraded to FULL

    move    #SBits,r0                        ;SBits array
    move    #JntSBits,r1                    ;Joint stereo SBits array
    move    #SBndSKF,r2                    ;SKFs array
    move    #JntSBSKF,r3                   ;Joint stereo SKFs array
    move    #polydta,r4                    ;addr of left chan poly analyzed data
    move    #JntPlAnal,r5                  ;addr of joint chan poly analyzed data

    move    y:<sibound,x0                    ;intensity stereo sub-band boundary
    move    y:<usedsb,a                      ;count of subbands used
    sub     x0,a      x0,n0                 ;number of sub-bands to shift
    ; and position into reg SBits array
    clr     b      x0,n1                    ;clear b register for accum
    ; and position into joint SBits array
    add     x0,b      ;x0,n4                ;step over SKF subbands by 1 of 3 pos
    ; and pos into reg poly samples array
    add     x0,b      ;x0,n5                ;step over SKF subbands by 2 of 3 pos
    ; and pos into joint poly samples array
    add     x0,b      ;step over SKF subbands by 3 of 3 pos
    move    b1,n2                             ;adjust SKFs array to intesnity sub
    move    b1,n3                             ;adj joint SKFs array to intesnity sub

    move    (r0)+n0                          ;update SBits addr to boundary sub-band
    move    (r1)+n1                          ;update Joint SBits addr to boundary sb
    move    (r2)+n2                          ;upd SBndSKF addr to boundary sub-band
    move    (r3)+n3                          ;upd Joint SBndSKF addr to boundary sb
    move    r4,y0                             ;save starting addr Left poly samples
    move    r5,y1                             ;save starting addr Joint poly samples

```

161

BAD ORIGINAL

```

move    #NUMSUBBANDS,n0          ;to shift the right channel SBits
move    n0,n1                    ;to shift the right channel JntSbits
move    #NUMSUBBANDS*NPARGROUP,n2 ;to shift the right channel SBndSKF
move    n2,n3                    ;to shift right channel Joint SBndSKF
move    n0,n4                    ;to shift to next Left poly sample
move    n0,n5                    ;to shift to next Joint poly sample

;shift for the number of subbands requiring the shift
1. the SBits are changed to the joint SBits
2. the scale factors are changed to the joint scale factors
3. the left channel Poly Samples are replaced with the averaged
   joint Poly Samples (left + right)/2

do      a,_xcde_69

;overlay the left and right SBits code

move    x:(r1+n1),x0              ;get right channel joint SBits code
move    x0,x:(r0+n0)              ;replace the regular right SBits code
move    x:(r1)+,x0                ;get left channel joint SBits code
move    x0,x:(r0)+                ;replace the regular left SBits code

;overlay the group of 3 scale factors per sub-band for left and right channels

do      #NPARGROUP,_xcde_62
move    x:(r3+n3),x0              ;get right channel group SKF joint
move    x0,x:(r2+n2)              ;repl right channel group SKF regular
move    x:(r3)+,x0                ;get left channel group SKF joint
move    x0,x:(r2)+                ;repl left channel group SKF regular

;end of SKF shift loop current sub-band
xcde_62
nop

xcde_69

;move the full stereo left channel samples up to the intensity sub-band
; boundary into the joint averaged samples ((left + right) / 2) array

do      y:<sibound,_xcde_699

move    y0,r4                    ;set current sub-band sample 0 position
move    y1,r5                    ;set current sub-band sample 0 position

;shift the 36 samples for this sub-band

do      #NUMPERSUBBAND*NPARGROUP,_xcde_66
move    x:(r4)+n4,x0              ;get left channel sample
move    x0,x:(r5)+n5              ;insert left sample into joint sample

xcde_66

;set-up the starting sample address for the next sub-band

move    y0,r4                    ;get current sub-band sample 0 position
move    y1,r5                    ;get current sub-band sample 0 position
move    (r4)+                    ;incr to next sub-band sample 0
move    (r5)+                    ;incr to next sub-band sample 0
move    r4,y0                    ;save starting addr
move    r5,y1                    ;save starting addr

```



;continue with coding of SBits, SKfs and sample data

\_xcde\_599

;before doing anything else further, do the scale factor checksums  
and store in the end of the previous frame

jsr setckskf

;now continue coding the current frame

```

move    #SBits,r0           ;SBits array
move    #SBIdx,r1           ;sub-band indicies
jsr      setsbits           ;set the sbits

move    #SBndSKF,r0         ;scale factors
move    #SBits,r1           ;SBits array
move    #SBIdx,r2           ;sub-band indices
jsr      setskf             ;set the scale factors

move    #SBPos,r3           ;sub-band allocated positions
move    #SBndSKF,r2         ;scale factors
move    #polydta,r0         ;to set addr right chan poly anal data
move    #INPCM,n0           ;offset to right chan poly analyzed data
move    #polydta,r1         ;addr of left chan poly analyzed data
move    (r0)+n0             ;addr of right chan poly analyzed data

```

;if doing joint, substitute the joint sample array for the left channel

```

jclr    #JOINT_FRAMING,y:<stereo,_xcde_169 ;Not Joint Framing
jset    #JOINT_at_FULL,y:<stereo,_xcde_169 ;Joint upgraded to FULL
move    #JntPlAna1,r1       ;addr of joint left chan poly anal data

```

\_xcde\_169

jsr setdata ;set the data

;if protection CRC checksum is included, do the checksum calculation  
; and insert it into the frame following the header info

```

jclr    #PROTECT,y:<stereo,_xcde_70 ;if 0, protection not applicable
jsr      setcrc                ;set the checksum

```

\_xcde\_70

```

jsr      setancdata           ;output ancillary data
jsr      bitsfree             ;flush remainder of bits to buffer
move     #-1,m6               ;restore r6 to linear buffer control

```

;signal to host

INTERRUPT\_HOST\_CD

jmp top ;wait for the next frame

end start

© 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.

```

; if SAMTYPE==SAM16K
;   define MAXCRITBND5_16K '21'

```

```

; noise masker positions
;cb

```

```
cb_16k
```

```

dc 6 ; index 0
dc 7 ; index 1
dc 7 ; index 2
dc 7 ; index 3
dc 7 ; index 4
dc 8 ; index 5
dc 8 ; index 6
dc 9 ; index 7
dc 10 ; index 8
dc 11 ; index 9
dc 13 ; index 10
dc 15 ; index 11
dc 18 ; index 12
dc 21 ; index 13
dc 26 ; index 14
dc 33 ; index 15
dc 39 ; index 16
dc 46 ; index 17
dc 55 ; index 18
dc 64 ; index 19
dc 78 ; index 20

```

```
endcb_16k
```

```

; noise masker geometric position
;g_cb

```

```
g_cb_16k
```

```

dc 1 ; index= 0, freq(Hz)= 15.6
dc 7 ; index= 1, freq(Hz)= 109.4
dc 15 ; index= 2, freq(Hz)= 234.4
dc 22 ; index= 3, freq(Hz)= 343.8
dc 29 ; index= 4, freq(Hz)= 453.1
dc 36 ; index= 5, freq(Hz)= 562.5
dc 44 ; index= 6, freq(Hz)= 687.5
dc 53 ; index= 7, freq(Hz)= 828.1
dc 62 ; index= 8, freq(Hz)= 968.8
dc 73 ; index= 9, freq(Hz)= 1140.6
dc 85 ; index= 10, freq(Hz)= 1328.1
dc 99 ; index= 11, freq(Hz)= 1546.9
dc 115 ; index= 12, freq(Hz)= 1796.9
dc 135 ; index= 13, freq(Hz)= 2109.4
dc 153 ; index= 14, freq(Hz)= 2468.8
dc 187 ; index= 15, freq(Hz)= 2921.9
dc 223 ; index= 16, freq(Hz)= 3484.4
dc 266 ; index= 17, freq(Hz)= 4156.3
dc 316 ; index= 18, freq(Hz)= 4937.5
dc 375 ; index= 19, freq(Hz)= 5859.4
dc 446 ; index= 20, freq(Hz)= 6968.8

```

164

BAD ORIGINAL



```
dc 512 ; end of list indicator

endg_cb_16k
;
endif

; (c) 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.

;
if SAMTYPE==SAM22K
;
define MAXCRITBND5_22K '22'

; noise masker positions
;cb
cb_22k
dc 5 ; index 0
dc 4 ; index 1
dc 5 ; index 2
dc 5 ; index 3
dc 6 ; index 4
dc 5 ; index 5
dc 6 ; index 6
dc 7 ; index 7
dc 7 ; index 8
dc 8 ; index 9
dc 10 ; index 10
dc 11 ; index 11
dc 12 ; index 12
dc 16 ; index 13
dc 19 ; index 14
dc 23 ; index 15
dc 29 ; index 16
dc 33 ; index 17
dc 40 ; index 18
dc 47 ; index 19
dc 56 ; index 20
dc 72 ; index 21
endcb_22k

; noise masker geometric position
;g_cb
g_cb_22k
dc 1 ; index= 0, freq(Hz)= 21.5
dc 5 ; index= 1, freq(Hz)= 107.7
dc 10 ; index= 2, freq(Hz)= 215.3
dc 15 ; index= 3, freq(Hz)= 323.0
dc 20 ; index= 4, freq(Hz)= 430.7
dc 26 ; index= 5, freq(Hz)= 559.9
dc 31 ; index= 6, freq(Hz)= 667.5
dc 38 ; index= 7, freq(Hz)= 818.3
dc 45 ; index= 8, freq(Hz)= 969.0
dc 52 ; index= 9, freq(Hz)= 1119.7
dc 61 ; index= 10, freq(Hz)= 1313.5
dc 72 ; index= 11, freq(Hz)= 1550.4
dc 83 ; index= 12, freq(Hz)= 1787.3
dc 97 ; index= 13, freq(Hz)= 2088.7
```

165

BAD ORIGINAL



```

dc 115 ; index= 14, freq(Hz)= 2476.3
dc 136 ; index= 15, freq(Hz)= 2928.5
dc 161 ; index= 16, freq(Hz)= 3466.8
dc 192 ; index= 17, freq(Hz)= 4134.4
dc 229 ; index= 18, freq(Hz)= 4931.1
dc 272 ; index= 19, freq(Hz)= 5857.0
dc 323 ; index= 20, freq(Hz)= 6955.2
dc 387 ; index= 21, freq(Hz)= 8333.3
dc 512 ; end of list indicator

endg_cb_22k
endif

; 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.

if SAMTYPE==SAM24K
define MAXCRITBND5_24K '23'

; noise masker positions
;cb

cb_24k
dc 4 ; index 0
dc 5 ; index 1
dc 4 ; index 2
dc 5 ; index 3
dc 5 ; index 4
dc 5 ; index 5
dc 5 ; index 6
dc 6 ; index 7
dc 7 ; index 8
dc 8 ; index 9
dc 8 ; index 10
dc 10 ; index 11
dc 12 ; index 12
dc 14 ; index 13
dc 18 ; index 14
dc 21 ; index 15
dc 26 ; index 16
dc 31 ; index 17
dc 37 ; index 18
dc 43 ; index 19
dc 51 ; index 20
dc 66 ; index 21
dc 96 ; index 22
endcb_24k

; noise masker geometric position
;g_cb

g_cb_24k
dc 1 ; index= 0, freq(Hz)= 23.4
dc 5 ; index= 1, freq(Hz)= 117.2
dc 9 ; index= 2, freq(Hz)= 210.9
dc 14 ; index= 3, freq(Hz)= 329.1
dc 19 ; index= 4, freq(Hz)= 445.3

```

166

BAD ORIGINAL

```
dc 24 ; index= 5, freq(Hz) = 562.5
dc 29 ; index= 6, freq(Hz) = 679.7
dc 34 ; index= 7, freq(Hz) = 796.9
dc 41 ; index= 8, freq(Hz) = 960.9
dc 48 ; index= 9, freq(Hz) = 1125.0
dc 56 ; index= 10, freq(Hz) = 1312.5
dc 65 ; index= 11, freq(Hz) = 1523.4
dc 76 ; index= 12, freq(Hz) = 1781.3
dc 89 ; index= 13, freq(Hz) = 2085.9
dc 105 ; index= 14, freq(Hz) = 2460.9
dc 125 ; index= 15, freq(Hz) = 2929.7
dc 148 ; index= 16, freq(Hz) = 3468.8
dc 176 ; index= 17, freq(Hz) = 4125.0
dc 210 ; index= 18, freq(Hz) = 4921.9
dc 250 ; index= 19, freq(Hz) = 5859.4
dc 297 ; index= 20, freq(Hz) = 6960.9
dc 355 ; index= 21, freq(Hz) = 8320.3
dc 435 ; index= 22, freq(Hz) = 10195.3
dc 512 ; end of list indicator

endg_cb_24k
;
; endif
;
; (c) 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; if SAMTYPE==SAM32K
;
; define MAXCRITBND5_32K '24'
; noise masker positions
;cb
cb_32k
dc 3 ; index 0
dc 4 ; index 1
dc 3 ; index 2
dc 3 ; index 3
dc 4 ; index 4
dc 4 ; index 5
dc 4 ; index 6
dc 5 ; index 7
dc 5 ; index 8
dc 5 ; index 9
dc 7 ; index 10
dc 7 ; index 11
dc 9 ; index 12
dc 11 ; index 13
dc 13 ; index 14
dc 16 ; index 15
dc 19 ; index 16
dc 24 ; index 17
dc 27 ; index 18
dc 32 ; index 19
dc 39 ; index 20
dc 49 ; index 21
dc 72 ; index 22
dc 129 ; index 23
```

167

BAD ORIGINAL



```
endcb_32k
```

```
; noise masker geometric position
;g_cb
```

```
g_cb_32k
```

```
dc 1 ; index= 0, freq(Hz)= 31.3
dc 3 ; index= 1, freq(Hz)= 93.8
dc 7 ; index= 2, freq(Hz)= 218.8
dc 10 ; index= 3, freq(Hz)= 312.5
dc 13 ; index= 4, freq(Hz)= 406.3
dc 17 ; index= 5, freq(Hz)= 531.3
dc 21 ; index= 6, freq(Hz)= 656.3
dc 26 ; index= 7, freq(Hz)= 812.5
dc 31 ; index= 8, freq(Hz)= 968.8
dc 36 ; index= 9, freq(Hz)= 1125.0
dc 42 ; index= 10, freq(Hz)= 1312.5
dc 49 ; index= 11, freq(Hz)= 1531.3
dc 57 ; index= 12, freq(Hz)= 1781.3
dc 67 ; index= 13, freq(Hz)= 2093.8
dc 79 ; index= 14, freq(Hz)= 2468.8
dc 93 ; index= 15, freq(Hz)= 2906.3
dc 111 ; index= 16, freq(Hz)= 3468.8
dc 132 ; index= 17, freq(Hz)= 4125.0
dc 157 ; index= 18, freq(Hz)= 4906.3
dc 187 ; index= 19, freq(Hz)= 5843.8
dc 222 ; index= 20, freq(Hz)= 6937.5
dc 266 ; index= 21, freq(Hz)= 8312.5
dc 326 ; index= 22, freq(Hz)=10187.5
dc 423 ; index= 23, freq(Hz)=13218.9
dc 512 ; end of list indicator
```

```
endg_cb_32k
```

```
; endif
```

```
; (c) 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
```

```
; if SAMTYPE==SAM44K
```

```
; define MAXCRITBND5_44K '24'
```

```
; noise masker positions
```

```
;cb
```

```
cb_44k
```

```
dc 2 ; index 0
dc 3 ; index 1
dc 2 ; index 2
dc 3 ; index 3
dc 2 ; index 4
dc 3 ; index 5
dc 3 ; index 6
dc 3 ; index 7
dc 4 ; index 8
dc 4 ; index 9
dc 5 ; index 10
dc 5 ; index 11
```

168

BAD ORIGINAL



```
dc 7 ; index 12
dc 7 ; index 13
dc 10 ; index 14
dc 12 ; index 15
dc 14 ; index 16
dc 17 ; index 17
dc 20 ; index 18
dc 23 ; index 19
dc 28 ; index 20
dc 36 ; index 21
dc 52 ; index 22
dc 93 ; index 23
```

endcb\_44k

```
; noise masker geometric position
;g_cb
```

g\_cb\_44k

```
dc 0 ; index= 0, freq(Hz)= 0.0
dc 2 ; index= 1, freq(Hz)= 86.1
dc 4 ; index= 2, freq(Hz)= 172.3
dc 7 ; index= 3, freq(Hz)= 301.5
dc 9 ; index= 4, freq(Hz)= 387.6
dc 12 ; index= 5, freq(Hz)= 516.8
dc 15 ; index= 6, freq(Hz)= 646.0
dc 18 ; index= 7, freq(Hz)= 775.2
dc 21 ; index= 8, freq(Hz)= 904.4
dc 25 ; index= 9, freq(Hz)= 1076.7
dc 30 ; index= 10, freq(Hz)= 1292.0
dc 35 ; index= 11, freq(Hz)= 1507.3
dc 41 ; index= 12, freq(Hz)= 1765.7
dc 48 ; index= 13, freq(Hz)= 2067.2
dc 56 ; index= 14, freq(Hz)= 2411.7
dc 67 ; index= 15, freq(Hz)= 2885.4
dc 80 ; index= 16, freq(Hz)= 3445.3
dc 96 ; index= 17, freq(Hz)= 4134.4
dc 114 ; index= 18, freq(Hz)= 4909.6
dc 136 ; index= 19, freq(Hz)= 5857.0
dc 161 ; index= 20, freq(Hz)= 6933.7
dc 193 ; index= 21, freq(Hz)= 8311.8
dc 236 ; index= 22, freq(Hz)=10163.7
dc 307 ; index= 23, freq(Hz)=13221.4
dc 512 ; end of list indicator
```

endg\_cb\_44k

```
; endif
```

```
; (c) 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
```

```
; if SAMTYPE==SAM48K
```

```
; define MAXCRITBND5_48K '24'
```

```
; noise masker positions
```

```
;cb
```

cb\_48k

169

BAD ORIGINAL

```
dc 2 ; index 0
dc 2 ; index 1
dc 3 ; index 2
dc 2 ; index 3
dc 2 ; index 4
dc 3 ; index 5
dc 3 ; index 6
dc 3 ; index 7
dc 3 ; index 8
dc 4 ; index 9
dc 4 ; index 10
dc 5 ; index 11
dc 6 ; index 12
dc 7 ; index 13
dc 9 ; index 14
dc 11 ; index 15
dc 13 ; index 16
dc 15 ; index 17
dc 18 ; index 18
dc 22 ; index 19
dc 26 ; index 20
dc 33 ; index 21
dc 48 ; index 22
dc 85 ; index 23
endcb_48k

; noise masker geometric position
;g_cb

g_cb_48k
dc 0 ; index= 0, freq(Hz)= 0.0
dc 1 ; index= 1, freq(Hz)= 46.9
dc 4 ; index= 2, freq(Hz)= 187.5
dc 6 ; index= 3, freq(Hz)= 281.3
dc 8 ; index= 4, freq(Hz)= 375.0
dc 11 ; index= 5, freq(Hz)= 515.6
dc 14 ; index= 6, freq(Hz)= 656.3
dc 17 ; index= 7, freq(Hz)= 796.9
dc 20 ; index= 8, freq(Hz)= 937.5
dc 23 ; index= 9, freq(Hz)= 1078.1
dc 27 ; index= 10, freq(Hz)= 1265.6
dc 32 ; index= 11, freq(Hz)= 1500.0
dc 37 ; index= 12, freq(Hz)= 1734.4
dc 44 ; index= 13, freq(Hz)= 2062.5
dc 52 ; index= 14, freq(Hz)= 2437.5
dc 62 ; index= 15, freq(Hz)= 2906.3
dc 74 ; index= 16, freq(Hz)= 3468.8
dc 88 ; index= 17, freq(Hz)= 4125.0
dc 104 ; index= 18, freq(Hz)= 4875.0
dc 124 ; index= 19, freq(Hz)= 5812.5
dc 148 ; index= 20, freq(Hz)= 6937.5
dc 177 ; index= 21, freq(Hz)= 8296.9
dc 217 ; index= 22, freq(Hz)= 10171.9
dc 282 ; index= 23, freq(Hz)= 13218.8
dc 512 ; end of list indicator

endg_cb_48k

; endif
```





```

    opt    fc
;
; (c) 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; \UXCODE\compval.asm
;
; This routine is used to compress a 3 sample triple.
; It examines the values of each of the 3 samples and finds an alternate
; sequence of values which approximates the original 3 samples. The alternate
; sequence changes only the least significant bit of one of the 3 samples.
;
; For example, if the samples are quantized to 3 steps and the values of
; the samples are 0, 0, and 0, the the resulting 5 bit number is
;
;     0*9 + 0*3 + 0 = 0
;
; The 0,0,0 sequence is mapped into the sequence 011 since it was determined
; that the sequence 0,0,0 has a very low probability of occurrence.
;
; The following tables have been done
;
;     3, 3 step values packed into 5 bits -> 4 bits
;     3, 5 step values packed into 7 bits -> 6 bits
;     3, 7 step values stored in 9 bits -> 8 bits
;
; In all cases, a tripllett of values is considered. An entire block could have
; the same reduction but the tables would be large. The current approach
; requires very little change to the ISO MUSICAM routine.
;
; The following must be done to incorporate the changes into MUSICAM.
;
;     1. Tell the bit allocator that the positions 1, 2 and 3 corresponding
;        to the above 3 tables now have 4, 6 and 8 bits respectively
;        for each tripllett instead of 5, 7 and 9 bits.
;
;     2. Add a table lookup to convert a 5, 7, 9 bit value into
;        a 4, 6, 8 bit value respectively.
;
; The similar thing must be done in the decoder.
;
; on entry:
;     n4 - set to the number of bits sent to setvalue for encoding
;     n0 - set with the normally coded triplet value sent to setvalue
;         (this is used as index into the proper table)
; on exit:
;     y0 - contains the compressed value to replace the normal one
;         sent to setvalue
;
; destroyed:
;     r0
;     a
;
; this save variable for exclusive use by compval only
;
; section highmisc
; xdef    compvalROSave
;
; org     xhe:
; stcompval_xhe
; compvalROSave    ds    1

```

171

BAD ORIGINAL

```
endcompval_xhe
endsec
```

```
section compress
xdef table1
xdef table2
xdef table3
```

```
org yhe:
```

```
stcompress_yhe
```

```
;table1 compresses values from indexed bit allocation position 1 values
```

```
table1
dc      0      ; 0
dc      0      ; 1
dc      0      ; 2
dc      1      ; 3
dc      2      ; 4
dc      3      ; 5
dc      1      ; 6
dc      2      ; 7
dc      3      ; 8
dc      4      ; 9
dc      5      ; 10
dc      5      ; 11
dc      6      ; 12
dc      7      ; 13
dc      8      ; 14
dc      9      ; 15
dc      9      ; 16
dc     10      ; 17
dc     11      ; 18
dc     12      ; 19
dc     13      ; 20
dc     11      ; 21
dc     12      ; 22
dc     13      ; 23
dc     14      ; 24
dc     14      ; 25
dc     14      ; 26
```

```
;table2 compresses values from indexed bit allocation position 2 values
```

```
table2
dc      0      ; 0
dc      0      ; 1
dc      1      ; 2
dc      7      ; 3
dc      7      ; 4
dc      0      ; 5
dc      0      ; 6
dc      1      ; 7
dc      1      ; 8
dc      1      ; 9
dc      2      ; 10
dc      2      ; 11
dc      3      ; 12
```

172

BAD ORIGINAL



dc	4	; 13
dc	4	; 14
dc	5	; 15
dc	5	; 16
dc	5	; 17
dc	5	; 18
dc	5	; 19
dc	5	; 20
dc	5	; 21
dc	5	; 22
dc	5	; 23
dc	5	; 24
dc	6	; 25
dc	6	; 26
dc	7	; 27
dc	8	; 28
dc	8	; 29
dc	9	; 30
dc	9	; 31
dc	10	; 32
dc	11	; 33
dc	11	; 34
dc	12	; 35
dc	13	; 36
dc	14	; 37
dc	15	; 38
dc	16	; 39
dc	17	; 40
dc	17	; 41
dc	18	; 42
dc	19	; 43
dc	19	; 44
dc	20	; 45
dc	20	; 46
dc	20	; 47
dc	20	; 48
dc	20	; 49
dc	21	; 50
dc	21	; 51
dc	22	; 52
dc	23	; 53
dc	23	; 54
dc	24	; 55
dc	25	; 56
dc	26	; 57
dc	27	; 58
dc	28	; 59
dc	29	; 60
dc	30	; 61
dc	31	; 62
dc	32	; 63
dc	33	; 64
dc	34	; 65
dc	35	; 66
dc	36	; 67
dc	37	; 68
dc	38	; 69
dc	39	; 70
dc	39	; 71
dc	40	; 72

173

BAD ORIGINAL



dc	41	; 73
dc	41	; 74
dc	42	; 75
dc	42	; 76
dc	42	; 77
dc	42	; 78
dc	42	; 79
dc	43	; 80
dc	43	; 81
dc	44	; 82
dc	45	; 83
dc	45	; 84
dc	46	; 85
dc	47	; 86
dc	48	; 87
dc	49	; 88
dc	50	; 89
dc	51	; 90
dc	51	; 91
dc	52	; 92
dc	53	; 93
dc	53	; 94
dc	54	; 95
dc	54	; 96
dc	55	; 97
dc	56	; 98
dc	56	; 99
dc	57	; 100
dc	57	; 101
dc	57	; 102
dc	57	; 103
dc	57	; 104
dc	57	; 105
dc	57	; 106
dc	57	; 107
dc	57	; 108
dc	57	; 109
dc	58	; 110
dc	58	; 111
dc	59	; 112
dc	60	; 113
dc	60	; 114
dc	61	; 115
dc	61	; 116
dc	61	; 117
dc	62	; 118
dc	62	; 119
dc	54	; 120
dc	54	; 121
dc	61	; 122
dc	62	; 123
dc	62	; 124

;table3 compresses values from indexed bit allocation position 3 values

table3	dc	22	; 0
	dc	22	; 1
	dc	0	; 2
	dc	1	; 3

174

BAD ORIGINAL

dc	2	; 4
dc	2	; 5
dc	7	; 6
dc	0000	; unused
dc	3	; 8
dc	3	; 9
dc	3	; 10
dc	1	; 11
dc	2	; 12
dc	2	; 13
dc	7	; 14
dc	0000	; unused
dc	8	; 16
dc	3	; 17
dc	3	; 18
dc	4	; 19
dc	5	; 20
dc	6	; 21
dc	7	; 22
dc	0000	; unused
dc	8	; 24
dc	8	; 25
dc	9	; 26
dc	10	; 27
dc	11	; 28
dc	12	; 29
dc	13	; 30
dc	0000	; unused
dc	14	; 32
dc	14	; 33
dc	15	; 34
dc	16	; 35
dc	17	; 36
dc	18	; 37
dc	18	; 38
dc	0000	; unused
dc	52	; 40
dc	19	; 41
dc	19	; 42
dc	20	; 43
dc	21	; 44
dc	21	; 45
dc	57	; 46
dc	0000	; unused
dc	58	; 48
dc	58	; 49
dc	19	; 50
dc	20	; 51
dc	21	; 52
dc	21	; 53
dc	57	; 54
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused



dc	22	; 64
dc	22	; 65
dc	23	; 66
dc	24	; 67
dc	25	; 68
dc	30	; 69
dc	30	; 70
dc	0000	; unused
dc	26	; 72
dc	26	; 73
dc	27	; 74
dc	28	; 75
dc	29	; 76
dc	30	; 77
dc	30	; 78
dc	0000	; unused
dc	31	; 80
dc	32	; 81
dc	33	; 82
dc	34	; 83
dc	35	; 84
dc	36	; 85
dc	37	; 86
dc	0000	; unused
dc	38	; 88
dc	39	; 89
dc	40	; 90
dc	41	; 91
dc	42	; 92
dc	43	; 93
dc	44	; 94
dc	0000	; unused
dc	45	; 96
dc	46	; 97
dc	47	; 98
dc	48	; 99
dc	49	; 100
dc	50	; 101
dc	51	; 102
dc	0000	; unused
dc	52	; 104
dc	53	; 105
dc	54	; 106
dc	55	; 107
dc	56	; 108
dc	57	; 109
dc	57	; 110
dc	0000	; unused
dc	58	; 112
dc	58	; 113
dc	59	; 114
dc	60	; 115
dc	61	; 116
dc	61	; 117
dc	57	; 118
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused



dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	62	; 128
dc	62	; 129
dc	63	; 130
dc	64	; 131
dc	65	; 132
dc	66	; 133
dc	66	; 134
dc	0000	; unused
dc	67	; 136
dc	68	; 137
dc	69	; 138
dc	70	; 139
dc	71	; 140
dc	72	; 141
dc	73	; 142
dc	0000	; unused
dc	74	; 144
dc	75	; 145
dc	76	; 146
dc	77	; 147
dc	78	; 148
dc	79	; 149
dc	80	; 150
dc	0000	; unused
dc	81	; 152
dc	82	; 153
dc	83	; 154
dc	84	; 155
dc	85	; 156
dc	86	; 157
dc	87	; 158
dc	0000	; unused
dc	88	; 160
dc	89	; 161
dc	90	; 162
dc	91	; 163
dc	92	; 164
dc	93	; 165
dc	94	; 166
dc	0000	; unused
dc	95	; 168
dc	96	; 169
dc	97	; 170
dc	98	; 171
dc	99	; 172
dc	100	; 173
dc	101	; 174
dc	0000	; unused
dc	102	; 176
dc	102	; 177
dc	103	; 178
dc	104	; 179
dc	105	; 180
dc	106	; 181
dc	106	; 182
dc	0000	; unused

177

BAD ORIGINAL

dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	107	; 192
dc	107	; 193
dc	108	; 194
dc	109	; 195
dc	110	; 196
dc	111	; 197
dc	118	; 198
dc	0000	; unused
dc	112	; 200
dc	113	; 201
dc	114	; 202
dc	115	; 203
dc	116	; 204
dc	117	; 205
dc	118	; 206
dc	0000	; unused
dc	119	; 208
dc	120	; 209
dc	121	; 210
dc	122	; 211
dc	123	; 212
dc	124	; 213
dc	125	; 214
dc	0000	; unused
dc	126	; 216
dc	127	; 217
dc	128	; 218
dc	129	; 219
dc	130	; 220
dc	131	; 221
dc	132	; 222
dc	0000	; unused
dc	133	; 224
dc	134	; 225
dc	135	; 226
dc	136	; 227
dc	137	; 228
dc	138	; 229
dc	139	; 230
dc	0000	; unused
dc	140	; 232
dc	141	; 233
dc	142	; 234
dc	143	; 235
dc	144	; 236
dc	145	; 237
dc	146	; 238
dc	0000	; unused
dc	140	; 240
dc	147	; 241
dc	147	; 242
dc	148	; 243

178

BAD ORIGINAL



dc	149	; 244
dc	150	; 245
dc	150	; 246
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	151	; 256
dc	151	; 257
dc	152	; 258
dc	153	; 259
dc	154	; 260
dc	155	; 261
dc	155	; 262
dc	0000	; unused
dc	156	; 264
dc	157	; 265
dc	158	; 266
dc	159	; 267
dc	160	; 268
dc	161	; 269
dc	162	; 270
dc	0000	; unused
dc	163	; 272
dc	164	; 273
dc	165	; 274
dc	166	; 275
dc	167	; 276
dc	168	; 277
dc	169	; 278
dc	0000	; unused
dc	170	; 280
dc	171	; 281
dc	172	; 282
dc	173	; 283
dc	174	; 284
dc	175	; 285
dc	176	; 286
dc	0000	; unused
dc	177	; 288
dc	178	; 289
dc	179	; 290
dc	180	; 291
dc	181	; 292
dc	182	; 293
dc	183	; 294
dc	0000	; unused
dc	184	; 296
dc	185	; 297
dc	186	; 298
dc	187	; 299
dc	188	; 300
dc	189	; 301
dc	190	; 302
dc	0000	; unused

179

BAD ORIGINAL



dc	191	; 304
dc	191	; 305
dc	192	; 306
dc	193	; 307
dc	194	; 308
dc	195	; 309
dc	195	; 310
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	200	; 320
dc	196	; 321
dc	196	; 322
dc	197	; 323
dc	198	; 324
dc	199	; 325
dc	199	; 326
dc	0000	; unused
dc	200	; 328
dc	201	; 329
dc	202	; 330
dc	203	; 331
dc	204	; 332
dc	205	; 333
dc	205	; 334
dc	0000	; unused
dc	206	; 336
dc	207	; 337
dc	208	; 338
dc	209	; 339
dc	210	; 340
dc	211	; 341
dc	211	; 342
dc	0000	; unused
dc	212	; 344
dc	213	; 345
dc	214	; 346
dc	215	; 347
dc	216	; 348
dc	217	; 349
dc	217	; 350
dc	0000	; unused
dc	218	; 352
dc	219	; 353
dc	220	; 354
dc	221	; 355
dc	222	; 356
dc	223	; 357
dc	223	; 358
dc	0000	; unused
ac	224	; 360
dc	224	; 361
dc	225	; 362
dc	226	; 363

180

BAD ORIGINAL



dc	227	; 364
dc	228	; 365
dc	228	; 366
dc	0000	; unused
dc	224	; 368
dc	224	; 369
dc	229	; 370
dc	229	; 371
dc	230	; 372
dc	231	; 373
dc	231	; 374
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	0000	; unused
dc	200	; 384
dc	233	; 385
dc	233	; 386
dc	234	; 387
dc	232	; 388
dc	199	; 389
dc	199	; 390
dc	0000	; unused
dc	200	; 392
dc	233	; 393
dc	233	; 394
dc	234	; 395
dc	235	; 396
dc	235	; 397
dc	205	; 398
dc	0000	; unused
dc	236	; 400
dc	236	; 401
dc	237	; 402
dc	238	; 403
dc	239	; 404
dc	239	; 405
dc	239	; 406
dc	0000	; unused
dc	240	; 408
dc	241	; 409
dc	242	; 410
dc	243	; 411
dc	244	; 412
dc	245	; 413
dc	245	; 414
dc	0000	; unused
dc	246	; 416
dc	246	; 417
dc	247	; 418
dc	248	; 419
dc	249	; 420
dc	250	; 421
dc	254	; 422
dc	0000	; unused



```

dc      246      ; 424
dc      251      ; 425
dc      251      ; 426
dc      252      ; 427
dc      253      ; 428
dc      254      ; 429
dc      254      ; 430
dc      0000     ; unused
dc      246      ; 432
dc      251      ; 433
dc      251      ; 434
dc      252      ; 435
dc      253      ; 436
dc      231      ; 437
dc      231      ; 438
dc      0000     ; unused

```

endcompress\_yhe

endsec

org phe:

compval

move r0,x:compvalR0Save ;save the register

;test the number of bits to choose the proper table:

; 4 bits - corresponds to table 1 with a 4 bit coded value

; 6 bits - corresponds to table 2 with a 6 bit coded value

; 8 bits - corresponds to table 3 with a 8 bit coded value

```

move    n4,a
move    #>4,y0
cmp     y0,a    #>6,y0
jeq     _cval_20
        ;test for table 1 first
        ;is table 1 chosen
        ; & set up for testing for table 2
        ;if eq, go set proper table address

cmp     y0,a
jeq     _cval_10
        ;is table 2 chosen
        ;if eq, go set proper table address

move    #table3,r0
jmp     _cval_30
        ;must be table 3, set its address

```

```

_cval_10
move    #table2,r0
jmp     _cval_30
        ;set address of table 2

```

```

_cval_20
move    #table1,r0
        ;set address of table 1

```

\_cval\_30

```

nop
move    y:(r0+n0),y0
move    x:compvalR0Save,r0
nop
rts
        ;return the compressed value
        ;restore the register

```

```

cpt      fc
;
; c: 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; \AUXCODE\setsyst.asm
;
; title      'Set the system word'
;
; This routine outputs the 20-bit system header information in to the frame.
; The frame header immediately follows the 12 bit sync word.
;
; on entry
;     r6 = current offset in output array
;     y:sc = shift count
;
; on exit
;     a = destroyed
;     b = destroyed
;     y0 = destroyed
;     y1 = destroyed
;     r4 = destroyed
;     n4 = destroyed
;
; include 'def.asm'
; include 'box_ctl.asm'
;
; org      phe:
;
setsyst
;bits 0 thru 3 of MUSICAM frame header:
;     0 = ID: high (1) or low (0) sampling rate
;     1-2 = '10' identifies frame as MPEG-ISO Layer II
;     3 = CRC-16 protected: YES (0) or NO (1)
;
;     move     #smplidbit,r0          ;get addr of high/low sample rate id
;     jclr     #PROTECT,y:<stereo,_syst_00 ;protection does not apply if 0
;     move     #>SYSTHDR_1_PROTECT,y0 ;bits 0-3 of frame header with CRC
;     jset     #0,y:(r0),_syst_10     ;if high sample rate id, continue
;     move     #>SYSTHDR_1_PROTECT_LOW,y0 ;replace with header id for low
;     jmp      _syst_10
;
;_syst_00
;     move     #>SYSTHDR_1_NO_PROT,y0 ;bits 0-3 of frame header with CRC
;     jset     #0,y:(r0),_syst_10     ;if high sample rate id, continue
;     move     #>SYSTHDR_1_NO_PROT_LOW,y0 ;replace with header id for low
;
;_syst_10
;output frame header bits 0 thru 3
;
;     move     #NSYSTHDR_1,n4          ;number of bits
;     jsr      <setvalue              ;output the value
;
;bits 4 thru 7 of MUSICAM frame header: bit rate set as per dip switches
;
;     move     y:bitrate,y0           ;bits 4-7 of frame header
;     move     #NBITRATE,n4          ;number of bits

```

```

; if a CDQ2000 split rate mono frame, switch bit rate in frame header
        jclr    #SPLIT_MONO_FRAME,y:<stereo,_syst_20
        move    y:splitrte,y0                ;bits 4-7 of frame header

_syst_20
; output frame header bits 4 thru 7
        jsr     <setvalue                    ;output the value

; bits 8 and 9 of MUSICAM frame header: sampling rate
        move    y:smplcde,y0                ;bits 8-9 of frame header
        move    #NSAMPLERATE,n4             ;number of bits
        jsr     <setvalue                    ;output the value

; bits 10 and 11 of MUSICAM frame header:
; 10 = padding bit: 0-no padding bits 1-8 padding bits
; 11 = privacy bit: as set by user
; test if the frame is padded or not with 8 added bits
        clr     a                            ;to initialize bits 10 and 11
        move    a,x:tstfrme                 ;temp variable to set the bits
        move    y:usediff,a                 ;tst if padded frame code needed
        tst     a                            ;see if frame not padded (a = 0)
        jeq     _syst_30                    ;the padding bit is already set to 0

; frame is padded with 8 additional bits
        bset    #1,x:tstfrme                 ;bit 10 set for padded frame

_syst_30
; set privacy bit as per user input
        TST_CLR_HEADER_BIT_11_CD,_syst_40   ;if not 0, continue
        bset    #0,x:tstfrme                 ;set the privacy bit

_syst_40
; output frame header bits 10 and 11
        move    x:tstfrme,y0                ;formatted bits
        move    #NSYSTHDR_2,n4              ;number of bits
        jsr     <setvalue                    ;output the value

; bits 12 and 13 of MUSICAM frame header: mode
; full stereo, joint stereo, dual channel or mono
        move    y:opfrtyp,y0                ;bits 12-13 of frame header
        move    #NFRAMETYPE,n4              ;number of bits
        jsr     <setvalue                    ;output the value

; bits 14 and 15 of MUSICAM frame header: mode extension
; stereo intensity sub-band bound
; (applicable only to a joint stereo frame)
        move    y:stintns,y0                ;bits 14-15 of frame header
        move    #NSTINTENSITY,n4            ;number of bits

```

184

BAD ORIGINAL



```

        jsr      <setvalue          ;output the value

;bits 16 thru 19 of MUSICAM frame header:
;      16 = copyright: YES (1) or NO (0)
;      17 = original/home: copy (0) or original (1)
;      18-19 = emphasis

        clr      a                  ;to initialize bits 16 thru 19
        move     a,x:tstfrme        ;temp variable to set the bits
        TST_CLR_HEADER_BIT_16_CD,_syst_50 ;if not set, continue
        bset     #3,x:tstfrme      ;set copyright bit

_syst_50
        TST_CLR_HEADER_BIT_17_CD,_syst_60 ;if not set, continue
        bset     #2,x:tstfrme      ;set original bit

_syst_60
        TST_CLR_HEADER_BIT_18_CD,_syst_70 ;if not set, continue
        bset     #1,x:tstfrme      ;set bit 1 of emphasis

_syst_70
        TST_CLR_HEADER_BIT_19_CD,_syst_80 ;if not set, continue
        bset     #0,x:tstfrme      ;set bit 0 of emphasis

_syst_80

;output frame header bits 16 thru 19

        move     x:tstfrme,y0      ;formatted bits
        move     #NSYSTHDR_3,n4    ;number of bits
        jsr      <setvalue          ;output the value

        rts

```

185

BAD ORIGINAL



```

; opt fo.mex
;
; c: 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; UXCODE\setdata.asm
;
; title 'Set the Data'
;
; This routine sets the data in the output buffer
;
; on entry
;   y:usedsb = number of used sub-bands
;   r3 = address of left & right channel SubBandPosition array (x memory)
;   r2 = address of left & right channel SubBandSKFs array (x memory)
;   r1 = address of the left channel poly analyzed data
;   r0 = address of the right channel poly analyzed data
;   y:opfrtyp = whether full stereo, joint stereo or mon frame
;   y:stereo = flags:
;       bit 0 means stereo vs mono framing
;           0 = stereo framing
;           1 = mono framing
;       bit 2 is to simply indicate that joint stereo applies
;           0 = NOT joint stereo framing type
;           1 = IS joint stereo framing type
;       bit 3 is to indicate the full stereo upgrade by allocate rtn
;           if joint stereo applies
;               0 = normal joint stereo allocation
;               1 = FULL STEREO allocation
;       bit 4 is to simply indicate the stereo intensity sub-band
;           boundary has been reached if joint stereo applies
;               0 = NO sub-bands still below intensity boundary
;               1 = sub-bands above intensity boundary
;   y:sibound = if joint stereo, sub-band boundary for stereo intensity
;
; on exit
;   a = destroyed
;   b = destroyed
;   x0 = destroyed
;   y0 = destroyed
;   x1 = destroyed
;   y1 = destroyed
;   r0 = destroyed
;   r2 = destroyed
;   r3 = destroyed
;   r4 = destroyed
;   r5 = destroyed
;   n5 = destroyed
;
; include 'def.asm'
; include 'box_ctl.asm'
; include '....\uxcode\quantize.mac'
; include '....\uxcode\setvalue.mac'
;
; section ytables
; xdef NBits,AA,BB
;
; org yhe:
; stsetdata_yhe

```



NB175

dc	0	; position = 0, place holder
dc	2	; position = 1
dc	3	; position = 2
dc	3	; position = 3
dc	4	; position = 4
dc	4	; position = 5
dc	5	; position = 6
dc	5	; position = 7
dc	7	; position = 8
dc	8	; position = 9
dc	9	; position = 10
dc	10	; position = 11
dc	11	; position = 12
dc	12	; position = 13
dc	13	; position = 14
dc	14	; position = 15
dc	15	; position = 16
dc	16	; position = 17

AA

dc	S000000	; position = 00, place holder
dc	S600000	; position = 01, .7500000000
dc	S500000	; position = 02, .6250000000
dc	S700000	; position = 03, .8750000000
dc	S480000	; position = 04, .5625000000
dc	S780000	; position = 05, .9375000000
dc	S7c0000	; position = 06, .9687500000
dc	S7e0000	; position = 07, .9843750000
dc	S7f0000	; position = 08, .9921875000
dc	S7f8000	; position = 09, .9960937500
dc	S7fc000	; position = 10, .9980468750
dc	S7fe000	; position = 11, .9990234380
dc	S7ff000	; position = 12, .9995117190
dc	S7ff800	; position = 13, .9997558590
dc	S7ffc00	; position = 14, .9998779300
dc	S7ffe00	; position = 15, .9999389650
dc	S7fff00	; position = 16, .9999694820
dc	S7fff80	; position = 17, .9999847410

BB

dc	S000000	; position = 00, place holder
dc	S600000	; position = 01, 1.0-.2500000000
dc	S500000	; position = 02, 1.0-.3750000000
dc	S700000	; position = 03, 1.0-.1250000000
dc	S480000	; position = 04, 1.0-.4375000000
dc	S780000	; position = 05, 1.0-.0625000000
dc	S7c0000	; position = 06, 1.0-.0312500000
dc	S7e0000	; position = 07, 1.0-.0156250000
dc	S7f0000	; position = 08, 1.0-.0078125000
dc	S7f8000	; position = 09, 1.0-.0039062500
dc	S7fc000	; position = 10, 1.0-.0019531250
dc	S7fe000	; position = 11, 1.0-.0009765630
dc	S7ff000	; position = 12, 1.0-.0004882810
dc	S7ff800	; position = 13, 1.0-.0002441410
dc	S7ffc00	; position = 14, 1.0-.0001220700
dc	S7ffe00	; position = 15, 1.0-.0000610350
dc	S7fff00	; position = 16, 1.0-.0000305180
dc	S7fff80	; position = 17, 1.0-.0000152590

187

BAD ORIGINAL



```
endsetdata_yhe
endsec
```

```
section highmisc
xdef sample1
xdef sample2
xdef sample3
```

```
org xhe:
stsetdata_xhe
```

```
sample1 ds 1 ;1st sample of a triplet
sample2 ds 1 ;2nd sample of a triplet
sample3 ds 1 ;3rd sample of a triplet
```

```
endsetdata_xhe
endsec
```

```
section highmisc
xdef blleft,blright,SKFaddr,POSaddr,bandcnt,block
xdef MaxiAdd,MaxiFact
```

```
org yhe:
stsetdata_yhe
```

```
blleft ds 1 ;left channel poly analyzed data
blright ds 1 ;right channel poly analyzed data
SKFaddr ds 1 ;save starting addr for SKF's
POSaddr ds 1 ;save starting addr for SBIndx's
bandcnt ds 1 ;incr sub-band for stereo intensity
block ds 1 ;block no 0:0-3, 1:4-7, 2:8-11
MaxiAdd ds 1 ;addr joint Maxi scale factors
MaxiFact ds 1 ;joint Maxi scale factor
```

```
endsetdata_yhe
endsec
```

```
org pli:
org phe:
```

```
setdata
```

```
move r2,y:SKFaddr ;save start address
move r3,y:POSaddr ;save start address

move r1,y:blleft ;save left channel start addr
move r0,y:blright ;save right channel start addr

move #NUMSUBBANDS,n1 ;spaced by number of subbands

move #0,r0 ;start group number
```

```
;loop through the 12 groups of 3 samples per sub-band per channel
;advancing through 36 samples
```

```
do #NUMPERSUBBAND,_setd_90
```

```
;set which block of SKF's (scale factor indices):
; 0 for group of 4 samples 0-3
; 1 for group of 4 samples 4-7
; 2 for group of 4 samples 8-11
```

188

BAD ORIGINAL

```

        move    r0,x0
        move    #>4,b           ;curr group to test
        cmp     x0,b    #>0,y1
        jgt     <_setd_00       ;block [0] groups 0 - 3

        move    #>8,b
        cmp     x0,b    #>1,y1
        jgt     <_setd_00       ;block [1] groups 4 - 7

        move    #>2,y1          ;block [2] groups 8 - 11
_setd_00
        move    (r0)+
        move    y1,y:block      ;increment the group number
                                   ;save which block[0, 1 or 2]

;set-up for joint stereo channel sub-band intensity control

        move    #JntSBMaxi,r4   ;addr of Joint Maxi factors
        move    y:sibound,n4    ;joint stereo intensity sub-band
        move    n4,y:bandcnt    ;bound subband decremented cntr
        rep     #NPERGROUP
        move    (r4)+n4
        move    r4,y:MaxiAdd    ;up JntMaxi table addr block 0
        bclr    #JOINT_at_SB_BOUND,y:<stereo ;save start of Joint Maxi facts
                                   ;clear reached boundary sub-band

;process for the defined used sub-bands this collection
; of three samples per sub-band per channel

        do      #NUMSUBBANDS,_setd_80

; if joint stereo does NOT apply, continue

        jclr    #JOINT_FRAMING,y:<stereo,_setd_08

; if joint stereo upgraded to full, continue

        jset    #JOINT_at_FULL,y:<stereo,_setd_08

; if doing joint stereo and have already switched over to joint SBits array,
; continue by getting the Maxi factor for the block

        jset    #JOINT_at_SB_BOUND,y:<stereo,_setd_05

; see if the joint stereo intensity sub-boundary has been reached

        move    r3,y:svereg     ;save reg 3
        move    y:bandcnt,r3    ;get decrement sub-band ctr
        jsr     chkjoint        ;see if reached boundary
        move    r3,y:bandcnt    ;save new decremented ctr
        move    y:svereg,r3     ;restore reg 3

; if intensity sub-band boundary NOT yet reached, continue

        jclr    #JOINT_at_SB_BOUND,y:<stereo,_setd_08
_setd_05

;get the Joint sub-band maxi factor for the group

```

```

move    r3,y:svereg           ;save reg 3
move    y:MaxiAdd,r3          ;get current Maxi sub-band
move    y:block,n3            ;which block for Maxi factor
nop
move    x:(r3+n3),y0          ;get the maxi factor
move    y0,y:MaxiFact         ;save for quantize routine
move    #NPERGROUP,n3         ;position to next sub-band
nop
move    (r3)+n3               ;adjust Maxi array addr to next
move    r3,y:MaxiAdd          ;save addr for next subband
move    y:svereg,r3           ;restore reg 3

_setd_08
move    y:blleft,r1           ;left channel block 1st
move    #0,n3                 ;left channel SBindx values
move    y:block,n2            ;which block of SKFs

;process left channel and then right channel for current sub-band
do      #NUMCHANNELS,_setd_75

;now, if doing the left channel, continue with outputting data
;otherwise, check for joint stereo and the intensity bound of sub-band
;if right channel joint stereo sub-band intensity boundary reached,
;  skip putting out the right channel value for this sub-band
;otherwise output the true right channel stereo values to the frame

jclr    #JOINT_at_SB_BOUND,y:<stereo,_setd_10 ;not joint boundary, go on
move    n3,b                  ;n3 is zero for left channel
tst     b                     ;test if doing left channel
jne     _setd_70              ;skip the right chan

_setd_10
move    #BB,r4                ;address of the B table
move    x:(r3+n3),n5           ;SubBandPosition[SubBand]
move    n5,a                  ;to test for NO index (0)
tst     a                     ;check position == 0 AND
                                ; set position for BValue fetch
                                ;none to output, try next chan
jeq     _setd_70

move    #AA,r5                ;address of the A table
move    y:(r4+n4),x1           ;BValue
move    y:(r5+n5),x0           ;AValue

move    #NBits,r5             ;address of NBits array
move    #>1,y0                ;test type of group
move    y:(r5+n5),n4           ;nbits

move    x:(r2+n2),n5           ;SKFIndex[SubBand] (block)
move    #IvSKF,r5             ;IvSKF table address

; test the position and pack those that qualify

cmp     y0,a    #>2,y0        ;check position == 1
jeq     <_setd_30
cmp     y0,a    #>4,y0        ;check position == 2
jeq     <_setd_40
cmp     y0,a    #>3,y0        ;check position == 4
jeq     _setd_50
cmp     y0,a
                                ;check pos == 3, and if not

```

```

;ine      <_setd_15                                ;handle all others not packed
; if not compressed mode, handle allocation position 3 normally
; if compression applies and NOT at the HIGH sampling rate,
;   handle allocation position 3 as a packed value

;set      #USE_COMPRESS y:<cmprsccl,_setd_45

; not position 1, 2, (3, if compression) or 4;
;   just a regular output of 3 adjacent data values

_setd_15
do        #NPERGROUP,_setd_20
move      x:(r1)+n1,y0                                ;get data value
;jsr      quantize                                    ;quantize the data
;MACRO: quantize the data
QUANTIZE
move      a1,y0                                ;move result into right reg
clr       a                                n4,b        ;set up a register for setvalue
;                                                ; & set len for setvalue macro
move      y0,a0                                ;set up for setvalue macro
;jsr      setvalue                                ;output the value
;MACRO: output the value
SETVALUE
nop

_setd_20
jmp       _setd_70

; Pos 1: Three adjacent data values are packed into 5 bits.
;   Each of the data values are only 2 bits wide.
;
;   packed_value = value0 * 9 + value1 * 3 + value2
;   or
;   packed_value = 3 * (value0 * 3 + value1) + value2

_setd_30
move      x:(r1)+n1,y0                                ;get 1st data value
move      y0,x:sample1
move      x:(r1)+n1,y0                                ;get 2nd data value
move      y0,x:sample2
move      x:(r1)+n1,y0                                ;get 3rd data value

;if new ISO CRC, also code CCS correction to packed values
; which switches the 1st and 3rd values in the triplet
; for ISO, 3rd value is correctly in place already in a register
; for CCS, save sample 3 and retrieve 1st sample into a register

;set      #CRC_OLD_vs_NEW,y:<stereo,_setd_31
move      y0,x:sample3
move      x:sample1,y0

_setd_31
;jsr      quantize                                    ;quantize the data
;MACRO: quantize the data
QUANTIZE
move      a1,b                                ;set to mult value by 3
lsl       b                                #0,a0        ;by 2
;                                                ; & kill extra bits
add       a,b                                x:sample2,y0 ;add for by 3 saving result in b

```

191

BAD ORIGINAL

```

; & get 2nd data value
; quantize the data
jsr    quantize
;MACRO: quantize the data
QUANTIZE
move    #0,a0
add     a,b
isl     b      b,a
;kill extra bits
;add 2nd to mult value by 3
;by 2
; & save total to add for by 3

;if new ISO CRC, also code CCS correction to packed values
; which switches the 1st and 3rd values in the triplet
; for ISO, 1st value is correctly in place already in a register
; for CCS, retrieve 3rd sample into a register

add     a,b      x:sample1,y0
;add for by 3 saving result in b
; & set sample 1 as 3rd sample
jset    #CRC_OLD_vs_NEW,y:<stereo,_setd_32
move    x:sample3,y0
;set- 3rd sample

_setd_32
jsr     quantize
;MACRO: quantize the data
QUANTIZE
add     b,a      #5,n4
;add in last result
; & nbits result for setvalue
move    a1,y0
;move to right register

;if compression applies:
; a. switch the bit count for setvalue
; b. set value for compression as register offset
; c. get the compressed value for setvalue

jclr    #USE_COMPRESS,y:<cmprctl,_setd_33
move    #4,n4
move    a1,n0
jsr     compval
;compress nbits for setvalue
;move to right register
;get compressed value

_setd_33
clr     a      n4,b
;set up a register for setvalue
; & set len for setvalue macro
;set up for setvalue macro
;output the value
move    y0,a0
jsr     setvalue
;MACRO: output the value
SETVALUE
jmp     _setd_70

; Pos 2: Three adjacent data values are packed into 7 bits.
; Each of the data values are only 3 bits wide.
;
; packed_value = value0 * 25 + value1 * 5 + value2
; or
; packed_value = 5 * (value0 * 5 + value1) + value2

_setd_40
move    x:(r1)+n1,y0
move    y0,x:sample1
move    x:(r1)+n1,y0
move    y0,x:sample2
move    x:(r1)+n1,y0
;get 1st data value
;get 2nd data value
;get 3rd data value

```

```

; if new ISO CRC, also code CCS correction to packed values
; which switches the 1st and 3rd values in the triplet
; for ISO, 3rd value is correctly in place already in a register
; for CCS, save sample 3 and retrieve 1st sample into a register

```

```

; set #CRC_OLD_vs_NEW, y: <stereo, _setd_41
move y0, x: sample3
move x: sample1, y0

```

```

_setd_41
; jsr quantize ; quantize the data
; MACRO: quantize the data
QUANTIZE
move a1, b ; set to mult value by 5
lsl b #0, a0 ; by 2
; & kill extra bits
lsl b ; by 4 (2 again)
add a, b x: sample2, y0 ; add for by 5 saving result in b
; & get 2nd data value
; quantize the data
; jsr quantize
; MACRO: quantize the data
QUANTIZE
move #0, a0 ; kill extra bits
add a, b ; add 2nd to mult value by 5
lsl b b, a ; by 2
; & save total to add for by 5
lsl b ; by 4 (2 again)

```

```

; if new ISO CRC, also code CCS correction to packed values
; which switches the 1st and 3rd values in the triplet
; for ISO, 1st value is correctly in place already in a register
; for CCS, retrieve 3rd sample into a register

```

```

add a, b x: sample1, y0 ; add for by 5 saving result in b
; & set sample 1 as 3rd sample
; set #CRC_OLD_vs_NEW, y: <stereo, _setd_42
move x: sample3, y0

```

```

_setd_42
; jsr quantize ; quantize the data
; MACRO: quantize the data
QUANTIZE
add b, a #7, n4 ; add in last result
; & nbits result for setvalue
move a1, y0 ; move to right register

```

```

; if compression applies:
; a. switch the bit count for setvalue
; b. set value for compression as register offset
; c. get the compressed value for setvalue

```

```

jclr #USE_COMPRESS, y: <cmprctl, _setd_43
move #6, n4 ; compress nbits for setvalue
move a1, n0 ; move to right register
jsr compval ; get compressed value

```

```

_setd_43
clr a n4, b ; set up a register for setvalue

```

193

```

; & set len for setvalue macro
; set up for setvalue macro
; output the value
move    y0,a0
jsr     setvalue
;MACRO: output the value
SETVALUE
jmp     _setd_70

_setd_45
; if compression applies for position 3:
; Pos 3: Three adjacent data values are packed into 8 bits.
; Each of the data values are only 3 bits wide.
;
; packed_value = value0 * 64 + value1 * 8 + value2
; or
; packed_value = 8 * (value0 * 8 + value1) + value2
;
move     x:(r1)+n1,y0          ;get 1st data value
jsr     quantize              ;quantize the data
;MACRO: quantize the data
QUANTIZE
move     a1,b                 ;set to mult value by 8
lsl      b                    ;by 2
lsl      b                    ;by 4 (2 again)
lsl      b                    ;by 8 (2 again) save result in b
; & get 2nd value
; quantize the data
jsr     quantize
;MACRO: quantize the data
QUANTIZE
move     #0,a0                ;kill extra bits
add      a,b                  ;add to total to mult value by 8
lsl      b                    ;by 2
lsl      b                    ;by 4 (2 again)
lsl      b                    ;by 8 (2 again) save result in b
; & get 3rd value
; quantize the data
jsr     quantize
;MACRO: quantize the data
QUANTIZE
add      b,a                  ;add in last result
; & nbits result for setvalue
; move to right register
; get compressed value
clr      a                    ;set up a register for setvalue
; & set len for setvalue macro
; set up for setvalue macro
; output the value
move     y0,a0
jsr     setvalue
;MACRO: output the value
SETVALUE
jmp     _setd_70

; Pos 4: Three adjacent data values are packed into 10 bits.
; Each of the data values are only 4 bits wide.
;
; packed_value = value0 * 81 + value1 * 9 + value2
; or
; packed_value = 9 * (value0 * 9 + value1) + value2
;
_setd_50
move     x:(r1)+n1,y0          ;get 1st data value
move     y0,x:samp1el

```



```

        move    x:(r1,-n1,y0                ;get 2nd data value
        move    y0,x:sample2
        move    x:(r1)+n1,y0                ;get 3rd data value

;if new ISO CRC, also code CCS correction to packed values
;  which switches the 1st and 3rd values in the triplet
;  for ISO, 3rd value is correctly in place already in a register
;  for CCS, save sample 3 and retrieve 1st sample into a register

        jset    #CRC_OLD_vs_NEW,y:<stereo,_setd_51
        move    y0,x:sample3
        move    x:sample1,y0

_setd_51
;      jsr      quantize                    ;quantize the data
;MACRO: quantize the data
QUANTIZE
        move    a1,b                        ;set to mult value by 9
        lsl     b          #0,a0            ;by 2
                                           ; & kill extra bits
        lsl     b                        ;by 4 (2 again)
        lsl     b                        ;by 8 (2 again)
        add     a,b          x:sample2,y0    ;add for by 9 saving result in b
                                           ; & get 2nd value
;      jsr      quantize                    ;quantize the data
;MACRO: quantize the data
QUANTIZE
        move    #0,a0                      ;kill extra bits
        add     a,b                        ;add 2nd to mult value by 9
        lsl     b          b,a             ;by 2
                                           ; & save total to add for by 9
        lsl     b                        ;by 4 (2 again)
        lsl     b                        ;by 8 (2 again)

;if new ISO CRC, also code CCS correction to packed values
;  which switches the 1st and 3rd values in the triplet
;  for ISO, 1st value is correctly in place already in a register
;  for CCS, retrieve 3rd sample into a register

        add     a,b          x:sample1,y0    ;add for by 9 saving result in b
                                           ; & set sample 1 as 3rd sample
        jset    #CRC_OLD_vs_NEW,y:<stereo,_setd_52
        move    x:sample3,y0

_setd_52
;      jsr      quantize                    ;quantize the data
;MACRO: quantize the data
QUANTIZE
        add     b,a          #10,n4         ;add in last result
                                           ; & nbits result for setvalue
        move    a1,y0                      ;move to right register
        clr     a          n4,b             ;set up a register for setvalue
                                           ; & set len for setvalue macro
        move    y0,a0                      ;set up for setvalue macro
;      jsr      setvalue                    ;output the value
;MACRO: output the value
SETVALUE

; We have just finished the current channel
; and since the left was 1st, set up for the right channel

```

```

_setd_70
move    y:blright,r1           ;now right channel block
move    #>NUMSUBBANDS*NPERGROUP,a ;move to SKFs for right channel
move    y:block,x0             ;get current block offset
add     x0,a    #NUMSUBBANDS,n3 ;add right chan offset, set
                                ; AND set adj to right SBPos
move    a1,n2                  ;offset register 2

```

; We have just finished both channels for a sub-band.  
 ; 1. adjust left and right poly analyzed sample pointers to next sub-band  
 ; 2. increment SBPos array pointer for next sub-band  
 ; 3. increment the SKFs array pointer over previous sub-band's 2nd & 3rd SKFs

```

_setd_75
move    #>1,x0                 ;incr left and right rcv'd samps
move    y:blleft,a             ;left address prev sub-band
add     x0,a    y:blright,b    ;adj left chan, get right chan
move    a,y:blleft             ;save left addr next sub-band
add     x0,b    (r3)+          ;adj right chan, incr SBPos ptr
move    #3,n2                  ;adj SKFs by 3
move    b,y:blright            ;save right addr next sub-band
move    (r2)+n2                ;next sub-band SKFs addr

```

```

_setd_80

```

; We have just finished a group of 3 samples per sub-band and we must  
 ; get set for the next group or 3 samples:  
 ; 1. adjust the left and right poly analyzed sample pointers for  
 ; the 2nd and 3rd samples in the group just finished  
 ; 2. restore the starting address of the SBPos array  
 ; 3. restore the starting address of the SKFs array  
 ; 4. restore joint stereo sub-band intensity boundary

```

move    #>NUMSUBBANDS*2,x0     ;adj over 2nd & 3rd samples
move    y:blleft,a             ;left address prev sub-band
add     x0,a    y:blright,b    ;adj left ptr, get right ptr
move    a,y:blleft             ;save left addr next group
add     x0,b    y:POSaddr,r3    ;adj right ptr, reset SBPos ptr
move    b,y:blright            ;save right addr next group
move    y:SKFaddr,r2           ;reset start SKF address

```

```

_setd_90
rts

```

```

opt      fc

; c) 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
; UXCODE\tstsine.asm

title    'Check Maskers for Sine'

; This is a routine to test the Tonals for the presence of a sine wave.
; on entry: r0 == addr of the Maskers structure array

include 'def.asm'

org      phe:
tstsine
    move    r0,x:<SvReg0          ;save addr of Maskers array

;set the frame counter and sine flag from the proper channel

    move    y:sincntlft,x0        ;start with the left channel
    move    y:sintstlft,x1        ;start with left channel test flag
    jclr    #LEFT_vs_RIGHT,y:<stereo,_tsin_00 ;if left, continue
    move    y:sincntrgt,x0        ;switch to the right channel
    move    y:sintstrgt,x1        ;switch to right channel test flag

_tsin_00

;set the working variables with the values for the proper channel

    move    x0,x:<sincnt          ;set the working frame count value
    move    x1,x:<sintest        ;set sine flag for proper channel

;start looking for a sine wave in the current channel

    move    #>MINDB,x0           ;minimum value
    move    x0,x:<maxtonal        ;set minimum value for max tonal
    move    x:<nmasker,b          ;number of maskers in array
    move    #>TONAL,x1           ;to match TONAL only 1st pass

;loop thru the maskers array looking for the highest tonal

do      b,_tsin_20
move    #MASKERSTYPE,n0         ;offset for type of masker
nop
move    x:(r0+n0),a              ;get curr masker's type
cmp     x1,a      x:<maxtonal,y0 ;check if it's a tonal
; & get set to compare to curr max
jne     _tsin_10                ;if not a TONAL, continue

;test the power vs last high tonal power

    move    #MASKERSPWDRDB,n0     ;offset to PowerDb
    nop
    move    x:(r0+n0),a           ;get TONAL PowerDb
    cmp     y0,a      #MASKERSBIN,n0 ;compare curr to last max TONAL value
; & get set to save bin # if higher
jle     _tsin_10                ;not a new higher PowerDb, continue

    move    a,x:<maxtonal         ;save new max tonal

```

```

        move    x:(r0+n0),y0          ;get the bin number
        move    y0,x:<maxbin

_tsin_10
        move    #MASKERSSIZE,n0      ;size of Masker structure
        nop
        move    (r0)+n0              ;advance to next Masker structure

_tsin_20
;now that we have the max tonal, test if another masker is within 30 Db

        move    #30/192.65,x0        ;subtract 30 Db from max tonal
        move    x:<maxtonal,a        ;get the max tonal PowerDb
        sub     x0,a    #>-1,x1      ;subtract off 30 Db
        ; * set 2nd sub-band NOT a sine to XCODE
        move    a,y1                 ;value to check against
        move    #>-1,x0              ;set 1st sub-band NOT a sine to XCODE
        move    x:<SvReg0,r0         ;address the Masker structure

;loop thru the maskers array looking for the highest tonal

        do      b,_tsin_40
        move    #MASKERSBIN,n0       ;offset to bin number
        move    x:<maxbin,y0         ;to see if this is selected as max
        move    x:(r0+n0),a         ;get bin number
        cmp     y0,a    #MASKERSPWDRDB,n0 ;check if selected as max
        ; & set offset to PowerDb
        jeq     _tsin_30            ;it's the selected one, continue

;test the power vs last high tonal power

        move    x:(r0+n0),a         ;get masker PowerDb
        cmp     y1,a                ;compare curr to max TONAL - 30 Db
        jle     _tsin_30            ;not a new higher PowerDB, continue

;if PowerDb is within 30 Db, it's NOT a sine wave, stop checking

        enddo
        jmp     _tsin_100

_tsin_30
        move    #MASKERSSIZE,n0     ;size of Masker structure
        nop
        move    (r0)+n0             ;advance to next Masker structure

_tsin_40
;to test consecutive frame count before declaring a sine wave in a channel

        move    #>SINE_FRAME_COUNT,y0

; set channel as a sine wave after ensuring the sine wave persists

        move    x:<sincnt,a
        cmp     y0,a    #>1,y0
        jge     _tsin_50

;count another frame set as a sine wave

```



```

        add     y0,a
        move    a,x:<sinent
        jmp     _tsin_900

_tsin_50

;now set channel as a sine wave

        bset    #LEFT_SINE_WAVE,x:<sinentest

;we have a sine wave, determine the two sub-bands with the sine wave

        move    x:<maxbin,b           ;get the bin number and divide by 16
        asr     b                     ;divide by 2
        asr     b                     ;divide by 4
        asr     b                     ;divide by 8
        asr     b                     ;divide by 16
        move    b,r0                 ;save the sub-band

;now see if this is the 1st sub-band to increment for 2nd sub-band
; OR is this the 2nd sub-band to decrement for 1st sub-band
;mask off all but the lower 4 bits of bin number

        move    x:<maxbin,b           ;get the bin number
        move    #>$F,x0              ;to mask off all but lower 4 bits
        and     x0,b #>8,x0          ;mask off bits
                                      ; & set to test for increment
        cmp     x0,b                 ;if greater, increment for 2nd sub-band
        jgt     _tsin_70

        move    r0,x1                ;this is the 2nd sub-band of the pair
        move    r0,b                 ;check if sub-band 0
        tst     b                     ;check for sub-band 0
        jeq     _tsin_60              ;if 0, 1st sub-band equals 2nd sub-band
        move    (r0)-                 ;set 1st sub-band as previous

_tsin_60
        move    r0,x0                ;insert the 1st sub-band of the pair
        jmp     _tsin_900

_tsin_70
        move    r0,x0                ;this is the 1st sub-band of the pair
        move    (r0)+                 ;set 2nd sub-band as next sequential
        move    r0,x1
        jmp     _tsin_900

_tsin_100

;determined as NOT a sine wave, see if previously set as a sine wave
; if channel was not defined as a sine wave, DONE!!

        jclr    #LEFT_SINE_WAVE,x:<sinentest,_tsin_900

;set consecutive count before declaring a sine wave in a channel

        move    #>SINE_FRAME_COUNT,y0

;see that the sine wave has stopped persisting for N frames

        move    x:<sinent,a

```

```
        tst     a        #>1,y0
        jeq     _tsin_110
;decrement another frame NOT as a sine wave

        sub     y0,a
        move    a,x:<sincnt
;restore previous found sub-bands

        jset    #LEFT_vs_RIGHT,y:<stereo,_tsin_105
;reset for the left channel of the pair

        move    y:strtsinlft,x0          ;left channel last found 1st sub-band
        move    y:endsinlft,x1          ;left channel last found 2nd sub-band

        jmp     _tsin_900                ;DONE!!!
_tsin_105
;reset for the right channel of the pair

        move    y:strtsinrgt,x0          ;right channel last found 1st sub-band
        move    y:endsinrgt,x1          ;right channel last found 2nd sub-band

        jmp     _tsin_900                ;DONE!!!
_tsin_110
;now clear the channel as a sine wave

        bclr    #LEFT_SINE_WAVE,x:<sintest
        jmp     _tsin_900                ;DONE!!!
_tsin_900
        rts
```



```
opt fc,mex,cex
; (c) 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
; \XCODE\trapcell.asm

title 'Trap Cells'

xcode dsp trapcell.asm

section trapcell

org p:$0
jmp >start

; IRQA:
; react to the frame time millisecond interval
; qtalloc interrupt (quit bit allocation) for bit allocation

org p:$8
jsr >irqa

; IRQB:
; react to the frame time millisecond interval
; timer interrupt (start XPSYCHO and XCODE of new frame)

org p:$a
jsr >irqb

; SSI receive data interrupt:
; copy in next input PCM value from A-to-D converter

org pli:$c
jsr <ssirec
nop

; SSI receive data interrupt with exceptions:
; copy in next input PCM value from A-to-D converter

org pli:$e
jsr <ssirece ;handle input channel pcm data exception
nop

; SSI transmit data interrupt:
; output the next encoded frame word from buffer

org p:$10
jsr <ssixmt
nop

; SSI transmit data interrupt with exceptions:
; output the next encoded frame word from buffer

org p:$12
jsr <ssixmte
nop

; SCI receive serial communications interrupt:
; input the next ancillary data byte
```

201

```
    org    p:S14
    jsr     <scirec
    nop

; SCI receive serial data interrupt with exceptions:
;   input the next ancillary data byte

    org    p:S16
    jsr     <scirece
    nop

; HOST COMMAND - 24: get the encoder switches host vector

    org    p:S24
    jsr     >hostvector_24

; HOST COMMAND - 26: get the encoder framing type host vector

    org    p:S26
    jsr     >hostvector_26

; HOST COMMAND - 28: get the encoder iso header host vector

    org    p:S28
    jsr     >hostvector_28

; HOST COMMAND - 2A: get the psycho table offset ID for a new parameter value

    org    p:S2a
    jsr     >hostvector_2A

; HOST COMMAND - 2C: update the psycho table with a new parameter value

    org    p:S2c
    jsr     >hostvector_2C

; HOST COMMAND - 2E: clean host vector buffer: read double buffer

    org    p:S2e
    jsr     >hostvector_2E

; HOST COMMAND - 30: indicate to the host that the encoder interrupts
;                   are on and functioning

    org    p:S30
    jsr     >hostvector_30

; unexpected interrupts

    org    p:S2
    jsr     >stack_error
    org    p:S1a
    jsr     >sciidle_line
    org    p:S1c
    jsr     >scitimer
    org    p:S3e
    jsr     >illegal_inst

endsec
```

202

BAD ORIGINAL





```

    opt fc
;
;  1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
;  JXCODE\ssirec.asm

    title 'SSI receive data interrupt handler'

    include 'def.asm'
    include 'box_ctl.asm'
    include '...\Common\ioequ.asm'

;these save variables for exclusive use by the ssirec interrupt handlers only

    section lowmisc
    xdef    ssirecR7Save
    xdef    ssirecM7Save

    org     yli:
stssirec_yli
    ssirecR7Save    ds     1
    ssirecM7Save    ds     1

endssirec_yli
    endsec

; SSI Receiver interrupt

    org     pli:

ssirec
    move     r7,y:<ssirecR7Save    ;save register
    move     m7,y:<ssirecM7Save    ;save register

;set up to receive this next input PCM data value

    move     y:<ipwptr,r7          ;curr input PCM data write pointer
    move     #PCMSIZE*2-1,m7      ;set as a mod buffer for both channels

;!!!12/14/94
    nop
;test for which channel is incoming and align the pointer if needed
;if it's a right channel value, capture it to current address in buffer
;if it's a left channel value,
; left channel values are stored on even buffer addresses the right channel
; is stored in the adjacent odd buffer address

;;      TST_SET_RIGHT_PCM_INPUT_XPS,_ssi_05    ;if low, its the right channel
;see if a left channel input PCM data buffer address realignment is needed

;;      jclr     #0,y:<ipwptr,_ssi_05    ;if addr already even, continue
;align odd buffer address to even for the left channel addresses
; NOTE: this alignment should occur only once during steady operation

;;      move     (r7)-                ;align for left channel values

;;_ssi_05

```

203

BAD ORIGINAL



12/14/94

;capture the new input PCM value and store in the buffer (properly aligned)

```
    movep    x:<<M_RX,x:(r7)-      ;input the current channel PCM value
    move     r7,y:<ipwptr          ; & advance to next channel position
    move     r7,y:<ipwptr          ;save addr for the input PCM value

    move     y:<ssirecR7Save,r7    ;restore register
    move     y:<ssirecM7Save,m7    ;restore register
    rti
```

; SSI Receiver interrupt with exceptions

```
ssirece
    move     r7,y:<ssirecR7Save    ;save register
    movep    x:<<M_SR,r7          ;clear the exception
    movep    x:<<M_RX,r7          ;eat the input the data
    move     y:<ssirecR7Save,r7    ;restore register
    rti
```

204

BAD ORIGINAL

```

    opt fc
; (c) 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
; \UXCODE\ssixmt.asm

    title 'SSI interrupt handler'

; xcode dsp ssixmt.asm

    include '...\common\ioequ.asm'

; these save variables for exclusive use by the ssixmt interrupt handlers only

    section lowmisc
    xdef      ssixmtR7Save
    xdef      ssixmtM7Save

    org      yli:
stssixmt_yli

ssixmtR7Save    ds      1
ssixmtM7Save    ds      1

endssixmt_yli
endsec

; SSI Transmitter interrupt

    org      pli:

ssixmt
    move     r7,y:<ssixmtR7Save
    move     m7,y:<ssixmtM7Save

    move     y:<oprptr,r7          ;get output read buffer pointer
    move     y:<outsize,m7        ;circular buffer (2 frames worth)
    nop
    movep    y:(r7)+,x:<<M_TX      ;output word for the rdecode
    move     r7,y:<oprptr          ;update output read buffer pointer

    move     y:<ssixmtM7Save,m7
    move     y:<ssixmtR7Save,r7

    rti

; SSI Transmitter interrupt with exceptions

ssixmte
    move     r7,y:<ssixmtR7Save

    movep    x:<<M_SR,r7          ;clear the exeption
    movep    x:(r7)+,x:<<M_TX      ;output the data

    move     y:<ssixmtR7Save,r7

    rti

```

205

opt fc.cex

1994. Copyright Corporate Computer Systems, Inc. All rights reserved.

\UXCODE\polyanal.asm

title 'Analysis Polyphase Filter'

This routine performs the polyphase analysis filter on an input data set of 32 samples.

The input data is assumed to be ordered so the oldest data is at higher addresses. Newer data is put in at the "left" of the old data.

Observe the following about the M's

k = 0..63

M[00][k] = M[31][k] even k

M[14][k] = M[17][k]

M[15][k] = M[16][k]

M[00][k] = -M[31][k] odd k

M[14][k] = -M[17][k]

M[15][k] = -M[16][k]

Thus the S's can be calculated with one half of the calculations as follows:

The original formula

$S(i) = \sum_{k=0..63} M(i,k) * Y(k) \quad i = 0..31$

Now using the symmetry of the M's

The new way first calculates the following 32 values

define YP[k] k=0..31 as follows

YP[ 0] = Y[ 0] + Y[32]

YP[ 2] = Y[ 2] + Y[30]

YP[ 4] = Y[ 4] + Y[28]

YP[ 6] = Y[ 6] + Y[26]

YP[ 8] = Y[ 8] + Y[24]

YP[10] = Y[10] + Y[22]

YP[12] = Y[12] + Y[20]

YP[14] = Y[14] - Y[18]

YP[16] = Y[16]

YP[18] = Y[34] - Y[62]

YP[20] = Y[36] - Y[60]

YP[22] = Y[38] - Y[58]

YP[24] = Y[40] - Y[56]

YP[26] = Y[42] - Y[54]

YP[28] = Y[44] - Y[52]

YP[30] = Y[46] - Y[50]

```

      YP[1] = Y[1] - Y[31]
      YP[3] = Y[3] - Y[29]
      YP[5] = Y[5] - Y[27]
      YP[7] = Y[7] - Y[25]
      YP[9] = Y[9] - Y[23]
      YP[11] = Y[11] - Y[21]
      YP[13] = Y[13] - Y[19]
      YP[15] = Y[15] - Y[17]
      YP[17] = Y[17] - Y[15]
      YP[19] = Y[19] - Y[17]
      YP[21] = Y[21] - Y[19]
      YP[23] = Y[23] - Y[21]
      YP[25] = Y[25] - Y[23]
      YP[27] = Y[27] - Y[25]
      YP[29] = Y[29] - Y[27]
      YP[31] = Y[31] - Y[29]

      i = 0..15
;old way      even(i) = sum(k=0,2,4,..62) M(i,k) * Y(k)
;new way      even(i) =      M(i, 0)*YP( 0) + M(i, 2)*YP( 2) +
                           M(i, 4)*YP( 4) + M(i, 6)*YP( 6) +
                           M(i, 8)*YP( 8) + M(i,10)*YP(10) +
                           M(i,12)*YP(12) + M(i,14)*YP(14) +
                           M(i,16)*YP(16) + M(i,18)*YP(18) +
                           M(i,20)*YP(20) + M(i,22)*YP(22) +
                           M(i,24)*YP(24) + M(i,26)*YP(26) +
                           M(i,28)*YP(28) + M(i,30)*YP(30) +
                           M(i,32)*YP(32) + M(i,34)*YP(34) +
                           M(i,36)*YP(36) + M(i,38)*YP(38) +
                           M(i,40)*YP(40) + M(i,42)*YP(42) +
                           M(i,44)*YP(44) + M(i,46)*YP(46) +
                           M(i,48)*YP(48) + M(i,50)*YP(50) +
                           M(i,52)*YP(52) + M(i,54)*YP(54) +
                           M(i,56)*YP(56) + M(i,58)*YP(58) +
                           M(i,60)*YP(60) + M(i,62)*YP(62)

;old way      odd(i) = sum(k=1,3,5,..63) M(i,k) * Y(k)
;new way      odd(i) =      M(i, 1)*YP( 1) + M(i, 3)*YP( 3) +
                           M(i, 5)*YP( 5) + M(i, 7)*YP( 7) +
                           M(i, 9)*YP( 9) + M(i,11)*YP(11) +
                           M(i,13)*YP(13) + M(i,15)*YP(15) +
                           M(i,17)*YP(17) + M(i,19)*YP(19) +
                           M(i,21)*YP(21) + M(i,23)*YP(23) +
                           M(i,25)*YP(25) + M(i,27)*YP(27) +
                           M(i,29)*YP(29) + M(i,31)*YP(31) +
                           M(i,33)*YP(33) + M(i,35)*YP(35) +
                           M(i,37)*YP(37) + M(i,39)*YP(39) +
                           M(i,41)*YP(41) + M(i,43)*YP(43) +
                           M(i,45)*YP(45) + M(i,47)*YP(47) +
                           M(i,49)*YP(49) + M(i,51)*YP(51) +
                           M(i,53)*YP(53) + M(i,55)*YP(55) +
                           M(i,57)*YP(57) + M(i,59)*YP(59) +
                           M(i,61)*YP(61) + M(i,63)*YP(63)

      S(i) = even(i) + odd(i)
      S(31-i) = even(i) - odd(i)

; Based on the above, the M array is stored in memory as follows:

      M[00][0] M[00][2] M[00][4] .. M[00][32] M[00][1] M[00][3] .. M[00][31]
      .
      .
      M[15][0] M[15][2] M[15][4] .. M[15][32] M[15][1] M[15][3] .. M[15][31]

; on entry

      r0(x) = address of the oldest input 32 PCM samples (32)
              newest data at higher address
      m0 = set properly

      m2 = 63 (mod 64 buffer)

      r3 = address of the next location in X array to place new data
      m3 = 511 (mod 512 buffer)

```



```

; r5(x) = address of the S output vector (32)
;
; !!! this routine leaves the m registers set like on entry
;
; The X buffer must be allocated so it can be a mod buffer (512).
; The Y buffer must be allocated so it can be a mod buffer (64).
;
; on exit
; r0 = updated (incremented) for next iteration.
; r3 = updated (decremented) to beginning of x array for next iteration
; r5 = updated to point to input S vector address + 32
;
; a = destroyed
; x0 = destroyed
; y0 = destroyed
; r0 = destroyed
; r1 = destroyed
; r2 = destroyed
; r4 = destroyed
; n1 = destroyed
; n2 = destroyed
; n3 = destroyed
; n4 = destroyed
; n5 = destroyed
; include 'def.asm'
;
; section polyanac
; xdef polyc
;
; org yli:
; stpolyc_yli
;
; include '..\xipsycho\polyc.asm'
;
; endpolyc_yli
; endsec
;
; section polyanam
; xdef polym
;
; org yhe:
; stpolym_yhe
;
; include '..\xipsycho\polym.asm'
;
; endpolym_yhe
; endsec
;
; org pli:
;
; paralyse
;
; First move the pcm data into the x vector.
; Remember that the oldest pcm data is at the highest address.
;
; do #31, poly15
; move x:(r0)+n0,x0
; move x0,x:(r3)-

```

\_polyc

```
move    x:(r0)-n0,x0
move    x0,x:(r3)
```

At this point, r3 should point the the first valid data in x. This address is the newest information. As r3 is incremented, it points to older data.

Now the data is in the proper place

Window all the X data by the C vector.

```
Z(i) = C(i) * X.i      i=0..511
C = r4
X = r3
```

compute the Y vector

```
Y(i) = sum(j=0..7) Z(i+64j)      i = 0..63
```

```
Y = r2
```

This version makes the observation that the Z vector is a temporary and thus Y can be computed as follows:

```
Y(i) = sum(j=0..7) [C(i+64j) * X(i+64j)]      i = 0..63
```

This saves the storage space for Z and the store and load associated with Z.

There is something curious about the C's. They possess a certian symmetry. The C's range from 0..511. If one thinks about a new quantity called E, where where the E's are defined

```
E[000] = C[000]
E[001] = C[064]
E[007] = C[448]
E[008] = C[001]
E[009] = C[065]
E[015] = C[449]
E[504] = C[063]
E[505] = C[127]
E[511] = C[511]
```

Now observe that

```
E[259-i] = -E[260+i]      for i = 0..251
```

This fact allows us to only store 256+16 of the E's. (In fact if we were really clever with the code, we should only have to store 256 + 12 E's). The polyc array is really the E values. The trick is to try to store as much of the polyc array in low memory (0..ff) as possible so the parallel move proceeds as fast as possible for the mac instructions.

```
move    #polyc,r4
move    #ybuf,r2
```

```
;get addr of C window
;set address of y buf
```

209

BAD ORIGINAL

```

move    #64,n3
move    #9,n4
;set skip factor
;set to skip back

do      #31,_poly20
clr     a      x:(r3)-n3,x0      y:(r4)-,y0      ;get first data
mac     x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
mac     x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
mac     x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
mac     x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
mac     x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
mac     x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
mac     x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
mac     x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
macr    x0,y0,a r3:-
; & position X for next
;save as new Y

move    a,x:(r2)-
_poly20
move    (r4)-n4
do      #31,_poly25
clr     a      x:(r3)-n3,x0      y:(r4)-,y0      ;get first data
mac     -x0,y0,a x:(r3)+n3,x0      y:(r4)-,y0      ;compute Z
mac     -x0,y0,a x:(r3)-n3,x0      y:(r4)-,y0      ;compute Z
mac     -x0,y0,a x:(r3)+n3,x0      y:(r4)-,y0      ;compute Z
mac     -x0,y0,a x:(r3)+n3,x0      y:(r4)-,y0      ;compute Z
mac     -x0,y0,a x:(r3)+n3,x0      y:(r4)-,y0      ;compute Z
mac     -x0,y0,a x:(r3)+n3,x0      y:(r4)-,y0      ;compute Z
mac     -x0,y0,a x:(r3)+n3,x0      y:(r4)-,y0      ;compute Z
mac     -x0,y0,a x:(r3)+n3,x0      y:(r4)-,y0      ;compute Z
macr    -x0,y0,a r3)+
; & position X for next
;save as new Y

move    a,x:(r2)-
_poly25
move    (r3)-
move    (r3)-n3
;adjust for next round

```

; The (r3)- and (r3)-n3 above is used to position r3 to the next empty position. This position is one before the beginning of the array. This is at a lower addr. This is the address for the NEXT new information.

; Lastly calculate the sub-band output (32 sub-bands)

```

;
; i = 0..15
; even(i) = see above
; odd(i) = see above
; S(i) = even(i) - odd(i)
; S(31-i) = even(i) - odd(i)
;
; S = r5,r1
; M = r4
; Y = r2
; a = even(i) sum
; b = odd(i) sum

```

; First calculate the YP array from the Y array.

```

move    #32,n4
move    r2,r4
move    r2,r1
move    r4)-n4
;set start address of YP array

```



```

;
; r1 now points to YP[1]
; r2 now points to Y[1]
; r4 now points to Y[32]
;

move    #2,n2
move    n2,n4
move    n2,n1
;set output buffer increment

move    x:(r4)-n4,x0
do #7,_poly26
move    x:(r2)-n2,a
add     x0,a      x:(r4)-n4,x0
move    a,x:(r1)+n1

_poly26
move    x:(r2)-n2,a
add     x0,a #18,n2
move    a,x:(r1)-n1
;now do the last one

Now r1 points to YP[16]
Now r2 points to Y[16]
Now r4 points to Y[16]

move    (r1)-n1
;set r1 to point to YP[13]

move    #46,n4
move    (r2)+n2
move    (r4)+n4

Now r1 points to YP[18]
Now r2 points to Y[34]
Now r4 points to Y[62]

move    #2,n4
move    n4,n2

move    x:(r4)-n4,x0
do #6,_poly27
move    x:(r2)+n2,a
sub     x0,a      x:(r4)-n4,x0
move    a,x:(r1)+n1

_poly27
move    x:(r2)+n2,a
sub     x0,a      #47,n2
move    a,x:(r1)+n1

Now r1 points to YP[18]
Now r2 points to Y[48]
Now r4 points to Y[48]

move    #17,n4
move    (r2)-n2
move    (r4)-n4

move    r2,r1
;set to YP[1]

Now r1 points to YP[1]
Now r2 points to Y[1]
Now r4 points to Y[31]

move    #2,n4

```

211

BAD ORIGINAL





```

        mac      x0,y0,a x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,a x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,a x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,a x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,a x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,a x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,a x:(r2)-n2,x0      y:(r4)-,y0
        macr     x0,y0,a (r2)-

; do odd sums

        clr      b              x:(r2)-n2,x0      y:(r4)-,y0
        rep      #15
        mac      x0,y0,b x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,b x:(r2)+n2,x0      y:(r4)+,y0
        mac      x0,y0,b x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,b x:(r2)+n2,x0      y:(r4)+,y0
        mac      x0,y0,b x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,b x:(r2)+n2,x0      y:(r4)+,y0
        mac      x0,y0,b x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,b x:(r2)+n2,x0      y:(r4)+,y0
        mac      x0,y0,b x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,b x:(r2)+n2,x0      y:(r4)+,y0
        mac      x0,y0,b x:(r2)-n2,x0      y:(r4)-,y0
        mac      x0,y0,b x:(r2)+n2,x0      y:(r4)+,y0
        macr     x0,y0,b (r2)-

; set y to start
        move     b,x0
        add      a,b      #16,n5
        sub      x0,a b,x:(r5)+
; save odd sum
; even - odd
; even - odd
; & save the sum data
; save the diff data

_poly30
        move     a,x:(r1)-

        move     (r5)+n5
; set for next pass

        rts

        page
        title    'Poly Analyze one super block'

; This routine poly-analyzes 36 blocks consisting of 32 samples each.

; on entry
;   r0 = starting address of a block of 1152 data points
;   m0 = set appropriately may be a mod buffer if needed)
;   r5 = starting address of the output buffer for results
;   m5 = set appropriately may be a mod buffer if needed)

; on exit
;   a = destroyed
;   b = destroyed
;   x0 = destroyed
;   y0 = destroyed
;   x1 = destroyed
;   y1 = destroyed
;   r0 = destroyed

```

213

BAD ORIGINAL



```

; r0 = destroyed
; r1 = destroyed
; r4 = destroyed
; r5 = destroyed
; r3 = destroyed
; r1 = destroyed
; r2 = destroyed
; r3 = destroyed
; r4 = destroyed
; r5 = destroyed

.org     phe:

polyanal
    move     =63,m2           ;set x buffer to mod 64
    move     =511,m3          ;mod 512 buffer
    do       =36,_poly_44     ;do entire super-block
    jsr      panalys:         ;filter the data
    move     =63,m2           ;set x buffer to mod 64
_poly_44
    move     =-1,m2           ;restore m2
    move     =-1,m3           ;restore m3

    rts

page
; This function initializes the polyphase filter.
; It turns of the interrupt system for 512 cycles so beware.

; on entry
;     r0 = address of the X buffer for the analysis filter

; on exit
;     r0 = destroyed
;     a  = destroyed

.org     phe:

polyaini
    clr      a
    rep      #512
    move     a,x:(r0)+

    rts

```

2/4

BAD ORIGINAL

```

opt    fo
;
;   1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
;   UKCODE\mem.asm

        title    'Relocatable Memory Declarations'

        include  'def.asm'

        section  phase21
        xdef     Tonals,Maskers

        org      lhe:
stphase21_lhe
Tonals   ds      MAXTONALS*TONALSSIZE          ;tonal array
Maskers  ds      (MAXTONALS+NUMMAXCRITBND5)*MASKERSSIZE ;masker array-1 chans

endphase21_lhe
        endsec

        section  phase2x
        xdef     GlbMsk
        xdef     Alising

        org      xhe:
stphase2x_xhe

GlbMsk   ds      MAXNMSKFREQS*2                ;global masking array
Alising  ds      MAXTONALS*ALIASSIZE*2         ;aliasing buffer

endphase2x_xhe
        endsec

        section  NoisePwr
        xdef     NoisePwr

        org      lhe:
stNoisePwr_lhe
NoisePwr ds      NUMMAXCRITBND5                ;noise array

endNoisePwr_lhe
        endsec

        section  b_i
        xdef     b_ii

        org      yhe:
stb_i_yhe
b_ii     ds      512

endb_i_yhe
        endsec

        section  xtables
        xdef     Thres10SLB
        xdef     ThresSLB

```

215

BAD ORIGINAL

```

        org     xhe:
stThr11513_xhe
;Threshold of hearing 11 db down
Thr11513      ds      510

endThr11513_xhe
stThr513_xhe

;standard Threshold of hearing
Thr513        ds      510

endThr513_xhe
endsec

section .tables
xdef     fmap_x
xdef     cb_15k
xdef     cb_22k
xdef     cb_24k
xdef     cb_32k
xdef     cb_44k
xdef     cb_48k
xdef     g_cb_16k
xdef     g_cb_22k
xdef     g_cb_24k
xdef     g_cb_32k
xdef     g_cb_44k
xdef     g_cb_48k
xdef     bereich
xdef     SubBandMap

        org     yhe:
stfmap_yhe

        include '...\uxcode\fmap.asm' ;frequency mapping

endfmap_yhe
stcb_yhe

        include '...\uxcode\cb.asm' ;noise tables

endg_cb_yhe
stbereich_yhe

        include '...\common\bereich.asm'

endbereich_yhe
stsbmap_yhe

        include '...\xpsycho\sbmap.asm' ;sub-band mapping

endsbmap_yhe

```



```

endsec

section codepass
xdef SBMSr
xdef SBMNRmax
xdef MNRval
xdef SBIndx
xdef SBPos
xdef AtLimit
xdef SvUsedSBs
xdef MNRsbc

org xhe:
stcodepass_xhe

; This array holds the MinMaskingDb - SubBandMax for each of the
; 64 left (0-31) and right (32-63) subbands
; another 32 (64-95) are included as the joint channel array for allocation
SBMSr ds NUMSUBBANDS*3 ;Mask to Signal ratio by sub-band

; This array holds the deallocation selection values:
; (MinMaskingDb - SubBandMax) - SNR(position at next lower index)
; for each of the 64 left (0-31) and right (32-63) subbands
SBMNRmax ds NUMSUBBANDS*2 ;Mask-to-Signal ratio + SNR(PrevPos)

; This array is for deallocation based on the least damage and has the
; sub-band values at the next lower position ordered over the 2 channel
; range of sub-bands. This array is paralled with the MNRsbc array below.
MNRval ds NUMSUBBANDS*2 ;table of ordered values (sub-band/chan)

; these arrays are dimension by *2 providing for the left channel
; followed by the right channel
SBIndx ds NUMSUBBANDS*2 ;sub-band index
SBPos ds NUMSUBBANDS*2 ;sub-band positions left & right

; flags set when a sub-band reaches its limit of allocation:
; (one per left channel for 32 subbands
; and one per right channel for 32 sub-bands)
; bit 0: set if below the global masking threshold
; bit 1: set if not used or fully allocated
AtLimit ds NUMSUBBANDS*2

; The SvUsedSBs array is for restoration prior to a required
; joint stereo allocation.
; It is the saved array for the counters for sub-bands with assigned indices
; If a sub-band starts out below the Global Masking Threshold it takes
; a certain number of consecutive frames before it is skipped. Until that
; count down (SUBBANDSCTDOWN) reaches zero, the sub-band will receive at
; least one allocation.
SvUsedSBs ds NUMSUBBANDS*2

; This array is for deallocation based on the least damage and has the
; control info identifying sub-band number and channel of the ordered values
; in the MNRval array above.

```

217

BAD ORIGINAL

```

MNRsbc ds NUMSUBBANDS*2 ;table of associated sub-bands/channel:
                                sub-bands 0-31 (bits 0 thru 4)
                                channel flagged by bit 6:
                                0 = left
                                1 = right

endcodepass_xhe
endsec

section arrays
xdef MinMskDb
xdef SBMaxDb
xdef SBits
xdef SBndSKF

org xhe:
starrays_xhe

; these arrays are dimension by *2 providing for the left channel
; followed by the right channel

MinMskDb ds NUMSUBBANDS*2 ;minimum masking level in slb's
SBMaxDb ds NUMSUBBANDS*2 ;the maximum in each subband

; these arrays are dimension by *2 providing for the left channel
; followed by the right channel

SBits ds NUMSUBBANDS*2 ;the S Bit array (scale factor type)
SBndSKF ds NUMSUBBANDS*NPERGROUP*2 ;sub-band scale factors

endarrays_xhe
endsec

section jntdata
xdef JntPlAnal
xdef JntSBits
xdef JntSBSKF
xdef JntSBMaxi

org xhe:
startjntdata_xhe

;these arrays are developed for handling joint stereo which is the
;combining of the left and right channel values

JntPlAnal ds INPCM ;joint averaged left + right samples
JntSBits ds NUMSUBBANDS*2 ;the S Bit array (for joint stereo)
JntSBSKF ds NUMSUBBANDS*NPERGROUP*2 ;scale factors (joint stereo)
JntSBMaxi ds NUMSUBBANDS*NPERGROUP ;joint Maxi scale factors

endjntdata_xhe
endsec

section highmisc
xdef lvSKF

org lhe:
stlvskf

```



```
include '...uxcode-divskf.asm'  
enddivskf  
endsec
```

219

BAD ORIGINAL



```
opt      dex
```

```
; 1993. Copyright Corporate Computer Systems, Inc. All rights reserved.
```

```
EXCODE$logpow.asm
```

```
title 'convert to power and discrambling'
```

```
; This routine converts the amplitude data to power.
; The input to this routine is output of the real 1024 point FFT. The
; FFT output is in scrambled order. The output of the FFT should be such
; that the first output point is dc. The next output point corresponds to 46
; Hz, the next to 92 Hz ....
```

```
; This routine arranges the output of the fft in normal order and computes the
; power (amplitude squared). The output of the fft corresponds to a real and
; an imaginary data point. The power is computed as follows.
```

```
      Pow(i) = Real(i) * Real(i) + Imag(i) * Imag(i)
```

```
; The real and imaginary parts of the data are stored in x and y memory
; respectively.
```

```
; Old versions of the psychoacoustic software ignored the dc value. Currently
; it is still that way. It should just 0 the dc value in the future.
```

```
; This routine reads its input from r0 and places the output at
; $800. The output address is hardwired!!!!
; See Ben if you want to change the output address.
```

```
org      pli:
```

```
logpow
```

```
; Also remember that the memory is external so it is better to
; fetch from x and y memory in sperate instructions if possible
```

```
on entry
```

```
      r0 = address of the data to convert to power
      index1 = address in y memory of descrambling table :
      index2 = address in y memory of descrambling table :
```

```
on exit
```

```
      a = destroyed
      b = destroyed
      x0 = destroyed
      y0 = destroyed
      n4 = destroyed
      n5 = destroyed
      n6 = destroyed
      r0 = destroyed
      r1 = destroyed
      r2 = destroyed
      r3 = destroyed
      r4 = destroyed
      r5 = destroyed
      r6 = destroyed
```

```
      move     =1,n5
      move     r0,-
      move     n5,n5
```

220

BAD ORIGINAL

```

move    #3,n0
move    =index1,r2          ;discrambling table 1
move    =index2,r3          ;discrambling table 2
move    n0,n1
lua     r0+ ,r1
move    #9fff,r6            ;calculate first two points

move    x:(r1),y0
move    y:(r0),x0
mpy     x0,x0,a              x:(r0)+n0,x0
mac     x0,x0,a
move    a,l:(r6)

mpy     y0,y0,b              y:(r1)+n1,y0
mac     y0,y0,b              #9fff,r6
move    #1,n0
move    b,l:(r6)

move    r0,r4
move    n0,n4

do      #8,_discram1          ;calculate the rest
do      n0,_discram2

move    x:(r0)+,x0            y:(r4)+,y0
mpy     x0,x0,a              y:(r2)+,r6
mac     y0,y0,a              x:(r0)+,x0      y:(r4)+,y0

move    y:(r3)+,r5
move    (r6)-n6

mpy     x0,x0,b              (r5)-n5
mac     y0,y0,b              a,l:(r6)
move    b,l:(r5)
_discram2

move    n0,b                  ; multiply n0 by 2
lsl     b
move    b1,n0

move    n0,n4
move    (r0)+n0
move    (r4)+n4
_discram1

rts
end

```



```

opt    is

; 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; XXXXCODE intquant.asm

title   'Joint Stereo Quantize Data'

; This routine is used to quantize the data using the Joint values
; MaxiFactor applicable to the current sub-band and block of 12 samples.
; The resulting data is right justified in the result register.

; This routine takes 64 - 2*number_of_bits cycles

; on entry
;   a = value
;   y:MaxiFact = the scale factor Maxi for sub-band and block
;   x0 = quantizing A value
;   x1 = quantizing B value
;   n4 = number of bits (1 - 16)
;   !!!!!!!!!!!!! n4 must never be 0 or negative !!!!!!!!!!!!!!!!!!!!!!!

; on exit
;   a1 = result
;
;   a2 = destroyed
;   a0 = destroyed
;   y0 = destroyed
;   y1 = destroyed
;   r4 = destroyed

include 'def.asm'

section highmisc
xdef    lshfttbl

org     yhe:
stjntquant_yhe

lshfttbl
dc      $000000      ;bits = 0, place holder
dc      $100000      ;bits = 1, shift left 23 bits
dc      $080000      ;bits = 2, shift left 22 bits
dc      $040000      ;bits = 3, shift left 21 bits
dc      $020000      ;bits = 4, shift left 20 bits
dc      $010000      ;bits = 5, shift left 19 bits
dc      $008000      ;bits = 6, shift left 18 bits
dc      $004000      ;bits = 7, shift left 17 bits
dc      $002000      ;bits = 8, shift left 16 bits
dc      $001000      ;bits = 9, shift left 15 bits
dc      $000800      ;bits = 10, shift left 14 bits
dc      $000400      ;bits = 11, shift left 13 bits
dc      $000200      ;bits = 12, shift left 12 bits
dc      $000100      ;bits = 13, shift left 11 bits
dc      $000080      ;bits = 14, shift left 10 bits
dc      $000040      ;bits = 15, shift left 09 bits
dc      $000020      ;bits = 16, shift left 08 bits

endjntquant_yhe
endsec

```

222

BAD ORIGINAL

```

; section :inquant ;new
; xdef :inquantize ;new

; org p:520 ;new
; org p:11 ;rev 1
; org p:he: ;rev 1

;inquantize
;move y:MaxiFact,y0 ;get the Maxi scale factor
;lst a =lshtbl,r4 ;see if dividend is negative
;lit _quan_10 ;it is

; - dividend and - divisor

;move y:(r4-n4),y1
;and #Sfe,ccr ;clear the carry bit
;rep n4 ;value/scalefactor
;div y0,a
;div y0,a ;one more div
;div y0,a ;one more div

;move a0,y0 ;get result to a reg
;mpy y0,y1,a #qstbl,r4 ;left justify
;jmp _quan_20

; - dividend and - divisor

_quan_10
;neg a y:(r4-n4),y1 ;make +
;and #Sfe,ccr ;clear the carry bit
;rep n4 ;value/scalefactor
;div y0,a
;div y0,a ;one more div
;div y0,a ;one more div

;move a0,y0 ;get result to a reg
;mpy -y0,y1,a #qstbl,r4 ;left justify

_quan_20
;move a0,a
;tfr x1,a a,y0
;mac x0,y0,a y:(r4+n4),y1 ;form quantized result
;asr a ;divide by 2
;move a,y0
;mpy y1,y0,a ;right justify the bits

_quan_30
;rts
;endsec ;new

```

```

; opt: fo
;
; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; XCODEB irq.asm

; title 'IRQ interrupt handler for frame time interval timer'
;
; XCODEB irq.asm: XPSYCHO and XCODEB combined dsp
;
; include 'def.asm'

; these save variables for exclusive use by interrupt irq handlers only

section highmisc
xdef    irqbR6Save
xdef    irqbN6Save
xdef    irqbM6Save
xdef    irqbR7Save
xdef    irqbN7Save
xdef    irqbM7Save

org     xhe:
start_irq_xhe

irqbR6Save    ds    1
irqbN6Save    ds    1
irqbM6Save    ds    1
irqbR7Save    ds    1
irqbN7Save    ds    1
irqbM7Save    ds    1

end_irq_xhe
endsec

org     phe:

; This little gem must be executed as a slow interrupt routine since
; it modifies the carry bit.

irqa
; bset    =0,y:<qtalloc
; rti

; This routines swaps the processing addresses for the next frame to be
; processed: first by the XPSYCHO code and then formatted by the XCODE code
; This routine must have a higher priority than the ssi routines
; so r7 won't be changed by the ssi routine.

; This interrupt occurs each time an output buffer is done. This occurs
; each:
; 14 ms for 48k sampling
; 16.12244899 ms for 44.1K sampling
; 16 ms for 32k sampling
; 48 ms for 24K sampling
; 52.24489796 ms for 22.05K sampling
; 72 ms for 16K sampling

irqb
move     r7,x:irqbR7Save    ;save the register
move     r7,x:irqbM7Save    ;save m7

```

224

BAD ORIGINAL

```

        move    n7,x:irgbN7Save    ;save n7
        move    r6,x:irgbR6Save    ;save the register

;since the T-MUSICAM frame pulse is aligned with the left channel pulse
;align the A-to-D converter input to left channel, if needed

        move    y:<ipwptr,r7      ;next address for input pcm data
        move    =INPCM*3,n7      ;parameter for 3 channel of values
        move    =PCMSIZE*3-1,m7   ;set as a mod buffer-2 channels

;if next address is to save a left channel value, (on an even address), continue
;else, adjust the odd address by 1 to make it an even address to start
;the next frame properly capture values aligned left and right channel

        jclr    =0,y:<ipwptr,_irgb_00 ;if aligned for left channel, continue
        move    r7)-              ;back up to even address
        move    r7,y:<ipwptr      ;save aligned addr for next frame input

_irgb_00

;set input PCM sample address to start poly analysis of the frame just captured

        move    r7)-n7            ;back to start of just completed frame
        move    r7,x:<polyst       ;set current frame input buffer address

;shift for next frame to encode with the poly analyzed data

        move    y:<frmstrt,r7      ;get current frame start to output
        move    y:<frmnext,r6      ;addr of next frame buffer to encode
;;        move    y:<outsize,m7     ;circular buffer (2 frames worth)
        move    r6,y:<frmstrt      ;swap next buffer to current to encode
;!!!h221
        move    #reedsolomon,r6    ;to see if reed solomon frames
        move    y:<outsize,m7      ;circular buffer (2 or 3 frames worth)
        move    y:<outmus,n7       ;length of a frame
        jclr    =0,y:(r6),_not_reed ;if not reed solomon, addr all set
        move    y:<frmnext,r7      ;addr of next frame buffer to encode
        nop
        move    r7)-n7            ;output frame ahead of one to be coded
_not_reed
;!!!h221
        move    r7,y:<oprptr       ;set where to start output read from
        move    r7,y:<frmnext      ;set next address for output
        move    (r7)-              ;set last word of current frame
        move    r7,y:<frmlast      ;save last word addr for block seq numb

        bset    #0,y:<timer        ;flip the timer sensed flag

        move    x:irgbR6Save,r6    ;restore the register
        move    x:irgbN7Save,n7    ;restore the n7
        move    x:irgbM7Save,m7    ;restore the m7
        move    x:irgbR7Save,r7    ;restore the register

        nop
        rti

;!!!debug unexpected interrupts

        org     phe:

```

stack\_error

nop  
nop  
nop  
rti

scallb\_line

nop  
nop  
nop  
rti

scallmer

nop  
nop  
nop  
rti

illegal\_inst

nop  
nop  
nop  
nop  
nop  
rti



```
    opt fc
;
; (c) 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; \UXCODE\hstvector.asm

    title 'Host Vector receive data interrupts handler'

    include '..\common\ioequ.asm'

; Host Vector Receiver interrupts
;     read in the vector
;     if the receive data register not empty,
;         read in the vector again

; save the host vectors for processing the switches next time

    section highmisc
    xdef     host24_word
    xdef     host26_word
    xdef     host28_word
    xdef     host2A_word
    xdef     host2C_word
    xdef     host2CA0Save
    xdef     host2CA1Save
    xdef     host2CA2Save
    xdef     host2CX0Save
    xdef     host2CR0Save
    xdef     host2CN0Save
    xdef     host2CM0Save
    xdef     host2E_word
    xdef     host30_word

    org      yhe:
sthstvector_yhe

host24_word    ds      1
host26_word    ds      1
host28_word    ds      1
host2A_word    ds      1
host2C_word    ds      1
host2CA0Save   ds      1
host2CA1Save   ds      1
host2CA2Save   ds      1
host2CX0Save   ds      1
host2CR0Save   ds      1
host2CN0Save   ds      1
host2CM0Save   ds      1
host2E_word    ds      1
host30_word    ds      1

endhstvector_yhe
endsec

    org      phe:

; get the encoder switches host vector

hostvector_24
```

```

    movep    x:<<_HRX,y:host24_word
    jset     #M_HRDF,x:<<M_HSR,hostvector_24

    rti

;get the encoder framing type host vector
hostvector_26

    movep    x:<<M_HRX,y:host26_word
    jset     #M_HRDF,x:<<M_HSR,hostvector_26

    rti

;get the encoder iso header host vector
hostvector_28

    movep    x:<<M_HRX,y:host28_word
    jset     #M_HRDF,x:<<M_HSR,hostvector_28

    rti

;get the psycho table offset ID for a new parameter value
hostvector_2A

    movep    x:<<M_HRX,y:host2A_word
    jset     #M_HRDF,x:<<M_HSR,hostvector_2A

    rti

;update the psycho table with a new parameter value
hostvector_2C

    movep    x:<<M_HRX,y:host2C_word
    jset     #M_HRDF,x:<<M_HSR,hostvector_2C

;save registers needed

    move     a0,y:host2CA0Save
    move     a1,y:host2CA1Save
    move     a2,y:host2CA2Save
    move     x0,y:host2CX0Save
    move     r0,y:host2CR0Save
    move     n0,y:host2CN0Save
    move     m0,y:host2CM0Save

;update the entry in the table

    move     #ptable,r0                ;address of the psycho parameter table
    move     #-1,m0                    ;set to a linear buffer

;see if table entry offset is valid (0 thru 31)

    move     y:host2A_word,a           ;get the table entry offset
    move     #0,x0                     ;to test for greater than 0
    cmp      x0,a    #>31,x0           ;see if less than 0
                                           ; & get set to test upper limit offset

```

```

        fil      _host_2C_100      ;if less than 0, ignore the value
        cmp      x0,a      a,n0    ;see if offset to big
        ;& set addr offset into the table
        jgt      _host_2C_100      ;if to big an offset, ignore the value

;insert the new table value into active table and update sample rate table

        move     y:host2C_word,x0    ;get the parameter value
        move     x0,y:(r0+n0)        ;insert into its table entry
        move     y:psychaddr,r0     ;address of sample rate psycho table
        nop
        move     x0,y:(r0+n0)        ;insert into sample rate table entry

_host_2C_100

;restore the registers

        move     y:host2CA0Save,a0
        move     y:host2CA1Save,a1
        move     y:host2CA2Save,a2
        move     y:host2CX0Save,x0
        move     y:host2CR0Save,r0
        move     y:host2CN0Save,n0
        move     y:host2CM0Save,m0

        rti

;clean host vector buffer: read double buffer

hostvector_2E

        movep    x:<<M_HRX,y:host2E_word
        movep    x:<<M_HRX,y:host2E_word
        jset     #M_HRDF,x:<<M_HSR,hostvector_2E

        rti

;indicate to the host that the encoder interrupts are on and functioning

hostvector_30

        movep    x:<<M_HRX,y:host30_word
        jset     #M_HRDF,x:<<M_HSR,hostvector_30

        rti

```

```

        opt      tex

; 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; UXC00DE\hanning.asm

        title    'hanning window'

; This function is used to apply a hanning window on the data.
; The window coefficients are symmetric, ie 0 = 1023, 1 = 1022, ...
; Thus the storage of the coeffs is only 512 points.

; The window is applied to the x data (real) and the y data (imag)
; is set to all zero.

; The input data is stored oldest data at the lowest memory location.
; To read the data in temporal order, you must increment the location
; counter.

; on entry
;     r0 = address of the oldest input data in x memory
;     n0 = 2 to offset by channel count
;     r1 = address of the output data (real in x memory, imag in y)
;     m0 = set to right value to read input data

; on exit

;     r0 = destroyed
;     r1 = destroyed
;     r4 = destroyed
;     a  = destroyed
;     x0 = destroyed
;     y0 = destroyed

        section hanning
        xdef      hcoef

        org      yhe:
sthcoef_yhe

        include   '..\xlpsycho\hanwin.asm'

endhcoef_yhe
endsec

hanning    org      pli:
        move      #hcoef,r4                                ;addr of window
        nop

        clr       a          x:(r0)+n0,x0      y:(r4)+,y0      ;window data
        do        #511,_han_10
        mpyr      x0,y0,a x:(r0)+n0,x0      y:(r4)+,y0      ;window data
        move      a,x:(r1)+
_han_10

        move      (r4)-
        mpyr      x0,y0,a x:(r0)-n0,x0      y:(r4)-,y0
        move      a,x:(r1)-

```



```
; do last 312
```

```
    do      #512, _han_30
    mpyr     x0,y0,a x:r0)-n0,x0    y:r4)-,y0    ;window data
    move     a,x:r1)-
_han_30
```

```
; zero the y part
```

```
    clr      a      r1)-
    do      #1024, _han_30
    move     a,y:r1)-
;_han_30
```

```
    rts
```



```

opt    io,mex

; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.

;WC002B getsws.asm

title   'Get encoder external switch settings'

; This routine is used to interpret the external switches on the box
; defined for the encoder

; on exit
; x:tstrate = raw bit rate input from the switches
; x:tstfrme = framing mode input from the switches
; x:tstband = sub-band width code input from the switches
; x:tstbaud = ancillary data baud rate input from the switches
; x:tstsel1 = application of line 1 select switch
; x:tstsel2 = application of line 2 select switch
; x:tstfrmt = frame communication formatting
; x:tstreed = Reed/Solomon encoding switch
; !!! x:tstbits = Reed/Solomon bit count to take from end of MUSICAM frames

; y:<not_appl = bit 4 set if any switches changed

; destroyed:
; register a

include 'def.asm'
include 'box_ctl.asm'

section highmisc
xdef    select1                ;current setting of line 1 select switch
xdef    select2                ;current setting of line 2 select switch
xdef    tstrate
xdef    tstsmpl
xdef    tstfrme
xdef    tstband
xdef    tstbaud
xdef    tstoccs
xdef    tstsel1
xdef    tstsel2
xdef    tstfrmt
xdef    tstreed
; !!! xdef    tstbits
xdef    clntbits

org     xhe:

stgetsws_xhe

select1    ds     1        ;current setting of line 1 select switch
select2    ds     1        ;current setting of line 2 select switch
tstrate    ds     1        ;raw bit rate input from the switches
tstsmpl     ds     1        ;raw sampling rate input from the switches
tstfrme     ds     1        ;raw framing mode input from the switches
tstband     ds     1        ;raw sub-band width code input from the switches
tstbaud     ds     1        ;raw ancil data baud rate input from switches
tstoccs     ds     1        ;MPEG-ISO (0) vs old CCS CDQ2000's (0)
tstsel1     ds     1        ;raw application of line 1 select switch
tstsel2     ds     1        ;raw application of line 2 select switch

```

232

BAD ORIGIN

```

tstfrmt      ds      1      ;raw frame communication formatting
tstreed      ds      1      ;Reed/Solomon encoding switch
;!!!tstbits  ds      1      ;Reed/Solomon bit cnt from end of MUSICAM frames
clntbits     ds      1      ;client code defined as per client specs

endgetsws_xhe
endsec

org phe:

getsws

bcir      =4,y:<not_appl ;indicate no changes initially

clr      a
move     a,x:tstrate
move     a,x:tstsmpl
move     a,x:tstfrme
move     a,x:tstband
move     a,x:tstbaud
move     a,x:tstoccs
move     a,x:tstsell
move     a,x:tstsel2
move     a,x:tstfrmt
move     a,x:tstreed
;!!! move     a,x:tstbits
move     a,y:trailbits
move     a,x:clntbits

;check the dip switches to determine frame bit rate
; and ancillary data application and data baud rate

;switches that define the framing bit rate

GET_BIT_RATE_CD

;switches that define the sampling bit rate

GET_SAMPLE_RATE_CD

;switches that define the mode of framing: Stereo, Mono, Joint Stereo

GET_FRAME_TYPE_CD

;switches that define the bit allocation sub-band width code

GET_BAND_WIDTH_CD

;switches that define the ancillary data baud rate

GET_BAUD_RATE_CD

;switches to set if selecting line 1 and/or line 2

GET_SELECTED_LINES_CD

;set client specified code for inclusion in frame

GET_CLIENT_CODES_CD

```

233

BAD ORIGINAL

;check for any changes in the control switches that would cause a restart

```

move    x:tstrate,y1          ;look for a change in framing rate
move    y:rawrate,a
cmp     y1,a    x:tstsmpl,y1   ;set up to test sampling rate
jne     _gsws_80
move    y:smpirte,a
cmp     y1,a    x:tstfrme,y1   ;set up to test framing mode
jne     _gsws_80
move    y:frmtype,a
cmp     y1,a    x:tstband,y1   ;set up to test band width code
jne     _gsws_80
move    y:bandwidth,a
cmp     y1,a    x:tstbaud,y1   ;set up to test ancillary data baud
jne     _gsws_80
move    y:baudrte,a
cmp     y1,a    x:tstoccs,y1   ;set up to test MPEG-ISO vs old CCS
jne     _gsws_80
move    y:oldccs,a
cmp     y1,a    x:tstsel1,y1   ;set up to test line 1 selection
jne     _gsws_80
move    x:select1,a
cmp     y1,a    x:tstsel2,y1   ;set up to test line 2 selection
jne     _gsws_80
move    x:select2,a
cmp     y1,a    x:tstfrmt,y1   ;set up to test framing format
jne     _gsws_80
move    y:frmformat,a
cmp     y1,a    x:tstreed,y1   ;set up to test Reed/Solomon switch
jne     _gsws_80
move    y:reedsolomon,a
cmp     y1,a
jeq     _gsws_90

_gsws_80
bset    #4,y:<not_appl          ;indicate changes in external switches

_gsws_90
rts

```

234

BAD ORIGINAL





```

opt    fo
;
; 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; UKCODE\flushfrm.asm
;
; This routine outputs zero bits to the frame buffer
;   up to the bit count passed in.
; If the bit count has been passed already, an error condition is returned
;
; on entry
;   r0 = bit count to be zeroed up to
;   r4 = addr of reed solomon flag to skip flushing out the frame
;   y:<bitscnt = count of bits output to frame so far by setvalue rtn
;
; on exit
;   a = destroyed
;   b = destroyed
;   y0 = destroyed
;   y1 = destroyed
;   r0 = destroyed
;   r4 = destroyed
;   n4 = destroyed
;
; include 'box_ctl.asm'
;
; org     phe:
;
flushframe
; unless these are reed solomon frames,
; in that case, skip over the frame flush at this point
;
;clr     a                                ;set for OK return
;jset    #0,y:(r4),_flsh_90              ;if reed solomon, skip flush
;
;_flsh_00
;pad 0 bits up to the bit count
;
;move    r0,a                            ;get bit count to end zero fill
;move    y:<bitscnt,y0                    ;get count of bits put into frame so far
;cmp     y0,a                            ;see if end bit position reached
;jeq     _flsh_90                        ;we've reached the end
;
;JUST IN CASE: see if the coded data is TOO LONG !!!!!!!!
;
;jlt     _FLSH_HELP                      ;IF WE OVERSHOT END OF FRAME ????????
;
;subtract the bits output so far
;
;sub     y0,a    #>16,y1                  ;subtract bitscnt from bit total
;                                              ; & set to zero 16 bits at a time
;
;see if a full 16 bits can be zeroed, else do remainder
;
;cmp     y1,a    a,n4                    ;see if full 16 bits fit
;                                              ; & set remainder bit len for setvalue
;jle     _flsh_10

```

235

BAD ORIGINAL

```
;full 16 bits can be zeroed
    move    y1,r4                ;set 16 bit length for setvalue
_flush_10
;output zero bits to the end of the frame
    move    #0,y0
    jsr     setvalue
;go back and see if more bits to zero
    jmp     _flush_00
_FLUSH_HELP
;ERROR!!! this case should not occur
    ON_BITALLOC_LED_CD          ;!!! error we've overshoot
    move    #>-1,a              ;indicate the error
;!!!debug: dump the frame in question (pull of the ';' from next line)
;    jsr     dumpdata
_flush_90
    rts
```

```

opt    fc

; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
; UKCODE\findnoise.asm

title   'find noise maskers'

; This routine is used to find the noise maskers. It does this by adding
; the power within a critical band.

; There will be MAXCRITBNDS values put into the NoiseDb array. The values
; are the power in watts.

; on entry
;   r1 = address of the power array (1 memory)
;   r2 = address address of the noise array (1 memory)
; on exit
;   a = destroyed
;   b = destroyed
;   b = destroyed
;   r1 = destroyed
;   r2 = destroyed
;   r3 = destroyed

include 'def.asm'

org     phe:

findnoise

    move     y:cb,r3                ;get the critical band boundaries
    do       y:<maxcritbnds,_find_90
    move     y:(r3)+,b              ;get the # of bins for crit band
    clr      a
    do       b,_find_80
    move     l:(r1)+,b
    add      b,a                    ;add the power

_find_80
    move     a,l:(r2)+              ;save the noise
_find_90
    rts

```



```

opt      fo.cex

; 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; XXXXXX\findskf.asm
;
; This routine is used to determine the scale factors for the blocks
; and subbands. The input poly phase data is assumed to be dimensioned
;
; PolyData(NUMBLOCKS) (NUMPERSUBBAND) (NUMSUBBANDS)
;
; The resulting scale factors are assumed to be dimensioned
;
; SubBandSCFs(NUMSUBBANDS) (NUMBLOCKS)
;
on entry
    r0 = polydata starting address
    r1 = SubBandSKFs starting address
;
on exit
    a = destroyed
    b = destroyed
    x0 = destroyed
    x1 = destroyed
    y0 = destroyed
    y1 = destroyed
    r0 = destroyed
    r1 = destroyed
    r3 = destroyed
    r4 = destroyed
    r5 = destroyed
    r6 = destroyed
    n3 = destroyed
    n4 = destroyed
    n5 = destroyed
    n6 = destroyed

include 'def.asm'

org      phe:
findskf
    move    #multbl,n4          ;get mult by 3 table address
    move    #lowtbl,n5          ;get lower table boundary address
    move    #upptbl,n6          ;get upper table boundary address
    move    #>5,b              ;lower boundary

    move    #NUMSUBBANDS,n3      ;get skip factor

    do      y:<usedsb,_find_90
    move     r0,r3

    do      #NPERGROUP,_find_80

;    find the max in the sub-band
; !!! the following only works if NUMPERSUBBAND is even !!!
; !!! should fix so assembler kicks out if NUMPERSUBBAND is odd !!!

    move     x:(r3)+n3,a          ;Maxi = *p-- (-- is by NUMSUBBANDS)
    move     x:(r3)+n3,x0        ;temp = *p-- (-- is by NUMSUBBANDS)
    do      #NUMPERSUBBAND-2:/2,_find_80

```

```

    cmpm    x0,a      x:r3-n3,x1      ;Maxi = temp
    ; & fetch next value to check
    tlt     x0,a      ;set new maximum if necessary
    cmpm    x1,a      x:r3-n3,x0      ;Maxi = temp
    ; & fetch next value to check
    tlt     x1,a      ;set new maximum if necessary
_find_20
    cmpm    x0,a      ;Maxi = temp
    ; & set val for lower limit
    tlt     x0,a      ;set new maximum if necessary
    cmpm    b,a      #62,r4
    tlt     _find_60

```

; !!! end even NUMPERSUBBAND specific code

```

    abs     a          #20,r4          ;form absolute value
    rep     #21        ;move max of 22 bits
    norm    r4,a       ;normalize

    move    r4,r6
    move    r4,r5
    move    y:(r6+n6),y0      ;get upper bounder

    cmp     y0,a      y:(r4+n4),r4    ;see if in upper 1/3
    ; & mult r4 by 3
    jge     _find_60      ;it is

    move    y:(r5+n5),y1      ;get lower boundry
    cmp     y1,a      (r4)+        ;see if in middle 1/3
    ; & add 1 to r4
    jge     _find_60      ;it is
    move    (r4)+        ;must be in lower 1/3

_find_60
    move    r4,x:(r1)+        ;*SubBandSKFs++

_find_80
    move    (r0)+            ;SubBand++

_find_90
    rts

```

© 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.

UKCODE\dbadd.asm

This routine add two numbers which are logs

$C = A - B$

$C = 10.0 * \log_{10}(10.0^{**}(A/10.0) - 10.0^{**}(B/10.0))$

Assume that A is bigger than B.

$C = A - 10.0 * \log_{10}(1.0 - 10.0^{**}((B-A)/10.0))$

The term

$10.0 * \log_{10}(1.0 - 10.0^{**}((B-A)/10.0))$

can be approximated by a table where A-B is the index into the table (after appropriate scaling). This works since A-B is always guaranteed to be positive.

The table is set to be in .5 dB increments with a maximum dB difference of 31.5 dB. If the difference is greater than 31.5 dB, then A is returned with nothing added.

Similarly if B is bigger.

on entry

a = a

x0 = B

on exit

b = C

b = destroyed

x0 = destroyed

r6 = destroyed

section ytables

xdef DbAddTbl\_3db

xdef DbAddTbl\_6db

org yhe:

DbAddTbl\_3db

dc	0.0156250	; 3.0103000, DbDif =	0.0000000
dc	0.0143647	; 2.7674916, DbDif =	-0.5000000
dc	0.0131788	; 2.5390189, DbDif =	-1.0000000
dc	0.0120666	; 2.3247408, DbDif =	-1.5000000
dc	0.0110269	; 2.1244260, DbDif =	-2.0000000
dc	0.0100580	; 1.9377592, DbDif =	-2.5000000
dc	0.0091379	; 1.7643486, DbDif =	-3.0000000
dc	0.0083242	; 1.6037356, DbDif =	-3.5000000
dc	0.0075543	; 1.4554046, DbDif =	-4.0000000
dc	0.0068452	; 1.3187948, DbDif =	-4.5000000
dc	0.0061939	; 1.1933105, DbDif =	-5.0000000
dc	0.0055971	; 1.0783324, DbDif =	-5.5000000
dc	0.0050516	; 0.9732279, DbDif =	-6.0000000

240

BAD ORIGINAL

dc	0.0045540	; 0.3773604, DbDif =	-6.5000000
dc	0.0041010	; 0.7900975, DbDif =	-7.0000000
dc	0.0036895	; 0.7102195, DbDif =	-7.5000000
dc	0.0033163	; 0.6389203, DbDif =	-8.0000000
dc	0.0029784	; 0.5738222, DbDif =	-8.5000000
dc	0.0026730	; 0.5149694, DbDif =	-9.0000000
dc	0.0023972	; 0.4618361, DbDif =	-9.5000000
dc	0.0021485	; 0.4139269, DbDif =	-10.0000000
dc	0.0019245	; 0.3707776, DbDif =	-10.5000000
dc	0.0017230	; 0.3319562, DbDif =	-11.0000000
dc	0.0015419	; 0.2970616, DbDif =	-11.5000000
dc	0.0013792	; 0.2657238, DbDif =	-12.0000000
dc	0.0012333	; 0.2376020, DbDif =	-12.5000000
dc	0.0011024	; 0.2123840, DbDif =	-13.0000000
dc	0.0009851	; 0.1897844, DbDif =	-13.5000000
dc	0.0008800	; 0.1695429, DbDif =	-14.0000000
dc	0.0007860	; 0.1514228, DbDif =	-14.5000000
dc	0.0007018	; 0.1352092, DbDif =	-15.0000000
dc	0.0006265	; 0.1207077, DbDif =	-15.5000000
dc	0.0005592	; 0.1077423, DbDif =	-16.0000000
dc	0.0004991	; 0.0961541, DbDif =	-16.5000000
dc	0.0004453	; 0.0858000, DbDif =	-17.0000000
dc	0.0003973	; 0.0765510, DbDif =	-17.5000000
dc	0.0003545	; 0.0682913, DbDif =	-18.0000000
dc	0.0003162	; 0.0609165, DbDif =	-18.5000000
dc	0.0002820	; 0.0543331, DbDif =	-19.0000000
dc	0.0002515	; 0.0484573, DbDif =	-19.5000000
dc	0.0002243	; 0.0432137, DbDif =	-20.0000000
dc	0.0002000	; 0.0385351, DbDif =	-20.5000000
dc	0.0001784	; 0.0343609, DbDif =	-21.0000000
dc	0.0001590	; 0.0306374, DbDif =	-21.5000000
dc	0.0001418	; 0.0273160, DbDif =	-22.0000000
dc	0.0001264	; 0.0243538, DbDif =	-22.5000000
dc	0.0001127	; 0.0217119, DbDif =	-23.0000000
dc	0.0001005	; 0.0193560, DbDif =	-23.5000000
dc	0.0000896	; 0.0172553, DbDif =	-24.0000000
dc	0.0000798	; 0.0153821, DbDif =	-24.5000000
dc	0.0000712	; 0.0137119, DbDif =	-25.0000000
dc	0.0000634	; 0.0122229, DbDif =	-25.5000000
dc	0.0000566	; 0.0108953, DbDif =	-26.0000000
dc	0.0000504	; 0.0097118, DbDif =	-26.5000000
dc	0.0000449	; 0.0086567, DbDif =	-27.0000000
dc	0.0000401	; 0.0077161, DbDif =	-27.5000000
dc	0.0000357	; 0.0068777, DbDif =	-28.0000000
dc	0.0000318	; 0.0061302, DbDif =	-28.5000000
dc	0.0000284	; 0.0054640, DbDif =	-29.0000000
dc	0.0000253	; 0.0048701, DbDif =	-29.5000000
dc	0.0000225	; 0.0043408, DbDif =	-30.0000000
dc	0.0000201	; 0.0038689, DbDif =	-30.5000000
dc	0.0000179	; 0.0034484, DbDif =	-31.0000000
dc	0.0000160	; 0.0030735, DbDif =	-31.5000000

endDbAddTbl\_3db

DbAddTbl\_6db

dc	0.0312500	; 5.0205999, DbDif =	0.0000000
dc	0.0299710	; 5.7741972, DbDif =	-0.5000000
dc	0.0287294	; 5.5349831, DbDif =	-1.0000000
dc	0.0275250	; 5.3029399, DbDif =	-1.5000000
dc	0.0263576	; 5.0780378, DbDif =	-2.0000000
dc	0.0252271	; 4.8602359, DbDif =	-2.5000000



dc	0.0241332	; 4.6494815, DbDif =	-3.0000000
dc	0.0230755	; 4.4457116, DbDif =	-3.5000000
dc	0.0220537	; 4.2488521, DbDif =	-4.0000000
dc	0.0210674	; 4.0588188, DbDif =	-4.5000000
dc	0.0201159	; 3.8755184, DbDif =	-5.0000000
dc	0.0191989	; 3.6998482, DbDif =	-5.5000000
dc	0.0183157	; 3.5286972, DbDif =	-6.0000000
dc	0.0174658	; 3.3649468, DbDif =	-6.5000000
dc	0.0166484	; 3.2074711, DbDif =	-7.0000000
dc	0.0158629	; 3.0561379, DbDif =	-7.5000000
dc	0.0151085	; 2.9108093, DbDif =	-8.0000000
dc	0.0143847	; 2.7713422, DbDif =	-8.5000000
dc	0.0136904	; 2.6375896, DbDif =	-9.0000000
dc	0.0130251	; 2.5094004, DbDif =	-9.5000000
dc	0.0123878	; 2.3866210, DbDif =	-10.0000000
dc	0.0117778	; 2.2690950, DbDif =	-10.5000000
dc	0.0111942	; 2.1566648, DbDif =	-11.0000000
dc	0.0106363	; 2.0491716, DbDif =	-11.5000000
dc	0.0101031	; 1.9464559, DbDif =	-12.0000000
dc	0.0095939	; 1.8483585, DbDif =	-12.5000000
dc	0.0091079	; 1.7547208, DbDif =	-13.0000000
dc	0.0086442	; 1.6653850, DbDif =	-13.5000000
dc	0.0082020	; 1.5801950, DbDif =	-14.0000000
dc	0.0077806	; 1.4989964, DbDif =	-14.5000000
dc	0.0073790	; 1.4216371, DbDif =	-15.0000000
dc	0.0069966	; 1.3479674, DbDif =	-15.5000000
dc	0.0066326	; 1.2778407, DbDif =	-16.0000000
dc	0.0062863	; 1.2111132, DbDif =	-16.5000000
dc	0.0059569	; 1.1476444, DbDif =	-17.0000000
dc	0.0056436	; 1.0872974, DbDif =	-17.5000000
dc	0.0053459	; 1.0299388, DbDif =	-18.0000000
dc	0.0050630	; 0.9754391, DbDif =	-18.5000000
dc	0.0047943	; 0.9236722, DbDif =	-19.0000000
dc	0.0045392	; 0.8745164, DbDif =	-19.5000000
dc	0.0042970	; 0.8278537, DbDif =	-20.0000000
dc	0.0040671	; 0.7835700, DbDif =	-20.5000000
dc	0.0038491	; 0.7415553, DbDif =	-21.0000000
dc	0.0036422	; 0.7017035, DbDif =	-21.5000000
dc	0.0034460	; 0.6639124, DbDif =	-22.0000000
dc	0.0032601	; 0.6280838, DbDif =	-22.5000000
dc	0.0030838	; 0.5941233, DbDif =	-23.0000000
dc	0.0029168	; 0.5619402, DbDif =	-23.5000000
dc	0.0027585	; 0.5314475, DbDif =	-24.0000000
dc	0.0026086	; 0.5025620, DbDif =	-24.5000000
dc	0.0024666	; 0.4752040, DbDif =	-25.0000000
dc	0.0023321	; 0.4492969, DbDif =	-25.5000000
dc	0.0022048	; 0.4247680, DbDif =	-26.0000000
dc	0.0020842	; 0.4015475, DbDif =	-26.5000000
dc	0.0019702	; 0.3795688, DbDif =	-27.0000000
dc	0.0018622	; 0.3587684, DbDif =	-27.5000000
dc	0.0017600	; 0.3390858, DbDif =	-28.0000000
dc	0.0016634	; 0.3204631, DbDif =	-28.5000000
dc	0.0015719	; 0.3028455, DbDif =	-29.0000000
dc	0.0014854	; 0.2861806, DbDif =	-29.5000000
dc	0.0014036	; 0.2704184, DbDif =	-30.0000000
dc	0.0013262	; 0.2555117, DbDif =	-30.5000000
dc	0.0012531	; 0.2414154, DbDif =	-31.0000000
dc	0.0011839	; 0.2280865, DbDif =	-31.5000000

endDbAddTbl\_fdb

292

BAD ORIGINAL



```

        endsec

;turned into a macro for qcalclog.asm
;;
;;      org      phe:
;;
;;DbAdd
;;      tfr      a,b      y:dbaddtbl,r6      ;assume A is biggest - so save A
;;                                          ; & get table base address
;;      sub      x0,a      #>$182,y0      ;form A-B
;;                                          ; & get scale factor
;;      jge      _db10
;;
;;      neg      a      x0,b      ;make difference a positive number
;;                                          ; & B was bigger - so save B
;;_db10
;;      move     a,x0
;;      mpyr     x0,y0,a #>$40,x0      ;form index into db table
;;                                          ; & get upper limit + 1
;;      cmp      x0,a      a1,n6      ;see if within table range
;;                                          ; & set index in proper register
;;      jge      _db20
;;
;;      rnd      b      y:(r6+n6),x0      ;rnd of b not really necessary
;;                                          ; & get table entry
;;      add      x0,b
;;
;;_db20
;;      rts

```

© 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.

UXCODE\dbadd.mac

; DbAdd routine turned into a macro for qcalcio.asm

no!ist

DBADD macro

```

    tfr    a,b      y:dbaddtbl,r6    ;assume A is biggest - so save A
                                ; & get table base address
    sub    x0,a      #>S192,y0      ;form A-B
                                ; & get scale factor
    jge    _db10
    neg    a          x0,b          ;make difference a positive number
                                ; & B was bigger - so save B
_db10
    move   a,x0
    mpyr   x0,y0,a    #>S40,x0      ;form index into db table
                                ; & get upper limit + 1
    cmp    x0,a      a1,n6          ;see if within table range
                                ; & set index in proper register
    jge    _db20
    rnd    b          y:(r6+r11,x0  ;rnd of b not really necessary
                                ; & get table entry
    add    x0,b
_db20
    endm
    list

```

244

BAD ORIGINAL

© 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.

UXCODE\fft.asm

This program originally available on the Motorola DSP bulletin board.  
It is provided under a DISCLAIMER OF WARRANTY available from  
Motorola DSP Operation, 6501 Wm. Cannon Drive W., Austin, Tx., 78735.

Radix 2, In-Place, Decimation-In-Time FFT (smallest code size).  
Test program

Last Update 10 Sep 86 Version 1.1

opt nomd,mex,cre,nocex

include '..\uxcode\fftr16b.asm'

define points '1024'

define data 'Sc00'

define coef 'S400'

define dacol 'SA00'

;hanning and fft buff  
;for sine and cosine table  
;for fft table

org pli:

fft

fftr16b points,data,coef,coef1,dacol

; clean up after the fft has done its work.  
; This makes the fft routine a "nice" routine.

move #-1,m0

move m0,m1

move m1,m2

move m2,m3

move m3,m4

move m4,m5

move m5,m6

rts

245

BAD ORIGINAL



```

opt    fc

; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; UCOCDE.exe.asm

        title    'CRC polynomial calculation'

; This routine computes the CRC of a string of bits in Y-Memory.
; It can use any arbitrary polynomial generator.
; It does this by simulating the hardware shift register implementation.
; This means that it does the calculation a bit at a time.

; X1 is set to indicate the number of bits the msb of the first data
; word is shifted. For example, setting x1 = 0 implies the the first
; data value is left justified in the word.

; on entry
;   r0 = Y-memory address of data array, msb of data is in msb of the words
;   r1 = number of bits to checksum
;   x0 = check sum seed value: 'ffff00' or '000000'
;   x1 = bit offset of the first data bit from the msb position (0-23)
;   y1 = CRC generator polynomial left justified ($800500 for CRC-16)

; on exit
;   a1 = CRC right justified 16-bit value
;
;   a0 = destroyed
;   a2 = destroyed
;   b  = destroyed
;   x0 = destroyed
;   x1 = destroyed
;   y0 = destroyed
;   y1 = destroyed

; register usage
;   y1 = CRC polynomial value
;   r0 = address of the next data word
;   r1 = number of bit left to work on
;   a1,a0 = current data value and data value + 1
;   x0 = accumulator (16 bit shift register - left justified)
;   b  = general temp

        include '...\common\def.asm'

        section lowmisc
        xdef      crcstrt

        org      yli:
store_yli

        crcstrt ds      1

; flag to control step over checksum
; bit 0 - set after up to checksum
; bit 1 - set after stepping over check

        endorg_yli
        endsec

        org      phe:

```

246

BAD ORIGINAL



```

_crc
    move    y:(r0)-,a1          ;get the first data word
    move    x1,b                ;get any offset bits
    tst     b                   ;see if need to adjust the input data
                                ; & get the second data word
                                ;no adjustment necessary
    leq     _crc_13
    rep     x1                  ;move the msb of data to top of a1
    asl     a                   ;shift into place

_crc_13
    move    #>24,b              ;compute the number of bits for first ip
    sub     x1,b                ;number of bits remaining in 1st word
                                ; & zero checksum itself avoidance ctl
                                ;sum remaining bits in 1st word
    move    b,x1                ;offset to the checksum stored
    move    #CRC_STORED_BIT_OFFSET,r2 ;msb of the accumulator
    move    #000000,y0

_crc_20
    do      x1,_crc_60
    jset    #1,y:<crcstrt,_crc_28 ;passed over checksum finished
    jset    #0,y:<crcstrt,_crc_26 ;stepping over checksum continues
    move    (r2)-
    move    r2,b                ;count the bits processed
    tst     b                   ;test r2 reached 0 yet
    jne     _crc_28             ;is last bit before checksum reached
    bset    #0,y:<crcstrt       ;no, continue summing early bits
    move    #NCRCBITS,r2        ;flag to skip over checksum
    jmp     _crc_28             ;bits in checksum to skip over
                                ;sum this last bit

_crc_26
    move    (r2)-
    move    r2,b                ;count the bits processed
    tst     b                   ;test r2 reached 0 yet
                                ;is last bit of checksum reached
                                ; & decrement the bit ctr (r1)
    jne     _crc_45             ;no, continue skipping over bits
    bset    #1,y:<crcstrt       ;flag checksum skipped over
    jmp     _crc_45             ;skip over this bit

_crc_28
    tfr     a,b                 ;save the current data in b
    eor     x0,b                ;look at the lsb of both data and accum
    and     y0,b                ;only the msb
    jpl     _crc_30             ;if +, then no subtraction necessary

    move    x0,b                ;get the accumulator
    asl     b                   ;and shift left 1 bit
                                ; & decrement the bit ctr r1
                                ;and subtract the polynomial generator

    eor     y1,b
    jmp     _crc_40

_crc_30
    move    x0,b                ;get the accumulator
    asl     b                   ;shift the accumulator one bit left
                                ; & decrement the bit ctr r1

_crc_40
    move    b1,x0               ;save as the new accumulator

```

```
_crc_45    asl      a      r1,b      ;shift the data one bit left
           ; & setup test # bits left to process
           tst      b      ;see if any left
           jgt      _crc_50
           enddo      ;all done

           move      #008000,y0      ;shift value
           mpy      x0,y0,a #>ffff,x0 ;right adjust the crc to lsb of a1
           and      x0,a      ;remove the debris
           move      #0,a0      ;remove the debris
           move      #0,a2      ;remove the debris

           rts

_crc_50    nop

_crc_60    move      #>24,x1      ;set next loop count
           move      y:(r0)-,a0    ;get the next word
           jmp      _crc_20

           end
```

```

    opt    fo

; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; XXXXXX zeropowe.asm

    title  'zero power'

; This function is used to zero the power around a tonal.

; on entry
;   r1 = address of the power array in memory
;   r2 = address of the tonal structure in memory
;   r3 = number of tonals in the tonal structure
;   r4 = address of the range table in memory

; on exit

;   a = destroyed
;   b = destroyed
;   x0 = destroyed
;   y0 = destroyed
;   r1 = destroyed
;   r2 = destroyed
;   r5 = destroyed
;   n1 = destroyed
;   n2 = destroyed
;   n4 = destroyed

    include 'def.asm'

    org    phe:

zeropowe
    move    r3,a                ;get number of tonals
    tst     a                   ;check if a good number
    jle     _zero_90

    move     #0,a                ;value to set power to

    move     #2,y0               ;shift right 6 bits

    move     #TONALSBIN,n2       ;get offset to the bin
    move     r1,r5               ;save starting position
    move     (r2)+n2             ;position to first bin

    move     #TONALSSIZE,n2      ;now get the size of structure
    nop

    do       r3,_zero_90
    move     x0,r2-n2,n1         ;get the next bin number

    move     n1,a                ;test for max bin number
    move     #490,x0             ;max bin number
    cmp      x0,a                ;see if at max
                                ; & clear a to zero values
    jlt     _zero_90

;bin at max. break out of loop and exit

```

249

```

        enddo
        jmp      _zero_90

_zero_30
;process bins not at max

        move     n1,x0

        mpy      x0,y0,b (r1)+n1      ;shift right 6 bits
        move     b1,n4                ;save the offset into rngtbl
        nop
        move     y:(r4+n4),n1          ;get the range
        move     n1,b                 ;save for later
        asl      b      #>1,x0
        add      x0,b      (r1)-n1    ;2 * range + 1

; use a do loop to keep interrupts alive.
; remember, a rep keeps interrupts off

        do       b, zero_10
        move     a,1:(r1)+
;zero the power

_zero_10
        move     r5,r1                ;restore starting array addr

_zero_90

        rts

```





```

opt    fo

; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
; UXC00BA\setckskf.asm

title   'Set scale factor CRC checksums'

; This routine has the 4 scale factor check sums that apply to the
; current frame. They are then stored in the end of the previous frame that
; was just coded. These values are the last 48 bits (12 each) in the
; MUSICAM frame. They may be followed by any client reserved bits and
; the CCS CDQ2000 block serial number when in combined mode.
; The check sums protect groups of scale factors by sub-band range:
;     1. sub-bands 0 thru 3
;     2. sub-bands 4 thru 7
;     3. sub-bands 8 thru 11
;     4. sub-bands 12 thru 31

; on exit
;   r0 = destroyed
;   r1 = destroyed
;   n1 = destroyed
;   r2 = destroyed
;   n2 = destroyed
;   r3 = destroyed
;   r4 = destroyed
;   a2 = destroyed
;   a1 = destroyed
;   b = destroyed
;   x0 = destroyed
;   y0 = destroyed
;   y1 = destroyed

include 'def.asm'

section highmisc
xdef    private
xdef    skfcrwd
xdef    skfrcbt
xdef    calskfck
xdef    sbctl5
xdef    skfcnt1
xdef    skfcnt2
xdef    skfcnt3
xdef    skfcnt4

org     xhe:
stsetckskf_xhe

private dc    1                ;header indication of application
; 0 not appl, 1 frame has checksums

skfcrwd      ds     1          ;word at frame end for next frame skf checksums
skfrcbt      ds     1          ;bit offset for next frame skf checksums

;calculated skf check sums

calskfck     ds     1          ;checksum 0 -1 initialize as none yet
              ds     1          ;checksum 1 -1 initialize as none yet

```



```

        ds      :      ;checksum 2  -1 initialize as none yet!
        ds      :      ;checksum 3  -1 initialize as none yet!

sect15
        ;table to control checksum:
        ; word one = starting sub-band
        ; word two = count of sub-bands

        ds      :      ;sub-band 0
skfent1  ds      4      ; thru sub-band 3
        ds      4      ;sub-band 4
skfent2  ds      4      ; thru sub-band 7
        ds      8      ;sub-band 8
skfent3  ds      4      ; thru sub-band 11
        ds      12     ;sub-band 12
skfent4  ds      10     ; thru sub-band 31 adjustable.

endsetckskf_xhe
        endSec

        org      phe:

;this routine calculates and stores the frame checksums
;if the private bit in the frame header is set to 1

setckskf

;if flag from frame header says skf checksums included (private bit = 1)
; retrieve the scale factor checksum from the frame and save it

        move     #private,r2      ;to test for appliaction
        nop
        jclr     #0,x:(r2),_sskfcr_900 ;if not appl, return

;if this is the 1st frame after a restart, can't store scale factor crc's
        jclr     #2,x:(r2),_sskfcr_900 ;it's 1st frame, return

;replaced uncoded scale factors with '63'

        move     #SBIndx,r0      ;array of assigned indexes
        move     #SBndSKF,r1     ;array of scale factors
        move     #NPERGROUP,n1   ;3 scale factors per sub-band

;go through all indexes for left channel and set any unused sub-bands

        do       #NUMSUBBANDS,_sskfcr_2
        move     x:(r0)+,a      ;get left channel sub-band index
        tst      a      #>63,x0 ;test for zero, set rplace value
        jne      _sskfcr_0      ;if used, adjust SBndSKF address

;replace 3 scale factors with 63

        move     x0,x:(r1)-      ;scale factor 1
        move     x0,x:(r1)-      ;scale factor 2
        move     x0,x:(r1)-      ;scale factor 3
        jmp      _sskfcr_1

_sskfcr_0

;scale factors should remain as is, bump up the address

```

252

BAD ORIGINAL

```

        move    r1, -n1

_sskfrcr_1
        nop

_sskfrcr_2
;go through all indexes for right channel and set any unused sub-bands
;unless we have a JOINT stereo frame. In that case, do through sibound

        move    y:opfrtyp,b          ;frame type to see if joint
        move    #>JOINT_STEREO,x1    ;joint stereo code
        move    #SBIndx,r2           ;array of left channel assigned indexes
        cmp     x1,b    #NUMSUBBANDS,n2 ;see if joint
        jne     _sskfrcr_3            ; & if case not, all right chan indexes
                                        ;not joint, do all right channel indexes

;we have a joint frame, use only right channel real indexes
; set up the offset to the left channel indexes

        move    y:<sibound,y0          ;get boundary sub-band count
        move    #>NUMSUBBANDS,a        ;total sub-bands per channel
        sub     y0,a    y:<sibound,n2  ;calc remaining sub-bands
                                        ; & set 1st loop sub-bands to test
        move    a,y0                  ;save the remainig sub-band count
        move    (r2)+n2                ;set addr to start left channel indexes

_sskfrcr_3
;set the required right channel indexes at zero

        do      n2,_sskfrcr_6
        move    x:(r0)+,a              ;get right channel sub-band index
        tst     a    #>63,x0           ;test for zero, set replace value
        jne     _sskfrcr_4            ;if used, adjust SBndSKF address

;replace 3 scale factors with 63

        move    x0,x:(r1)+            ;scale factor 1
        move    x0,x:(r1)+            ;scale factor 2
        move    x0,x:(r1)+            ;scale factor 3
        jmp     _sskfrcr_5

_sskfrcr_4
;scale factors should remain as is, bump up the address

        move    (r1)+n1

_sskfrcr_5
        nop

_sskfrcr_6
;if doing a joint frame right channels, do the rest based on the left channel
; indexes

        cmp     x1,b    r2,r0          ;see if joint
        jne     _sskfrcr_9            ; & set the left channel start address
                                        ;not joint, done

```



;we have a joint frame, use only right channel real indexes  
 ;set up the offset to the left channel indexes  
 ;set the required right sub-bands based on left channel indexes at zero

```
do      y0,_sskfcrc_5
move    x0,r0,-,a      ;get left channel sub-band index
lsl     a      #>63,x0  ;test for zero, set replace value
jne     _sskfcrc_7      ;if used, adjust SBndSKF address
```

;replace 3 scale factors with 63

```
move    x0,x:r1,-      ;scale factor 1
move    x0,x:r1,-      ;scale factor 2
move    x0,x:r1,-      ;scale factor 3
jmp     _sskfcrc_8
```

\_sskfcrc\_7

;scale factors should remain as is, bump up the address

```
move    r1,r1+1
```

\_sskfcrc\_8

```
nop
```

\_sskfcrc\_9

;initialize the sub-band counts for the 4 CRC-checksums

```
move    #>4,x0      ;set sub-band cnt for 1st 3 groups
move    x0,x:skfcnt1 ;set sub-band count group 1
move    x0,x:skfcnt2 ;set sub-band count group 2
move    x0,x:skfcnt3 ;set sub-band count group 3
```

;now make any adjustments if applicable MAXSUBBANDS is 12 or less

```
move    y:<maxsubs,a ;get applicable MAXSUBBANDS or testing
move    #>12,x0      ;total sub-band cnt for 1st 3 groups
cmp     x0,a      #>0,x1 ;see if more than 12 MAXSUBBANDS
; & set to zero subs count if needed
jgt     _sskfcrc_10 ;if so, go to set sub-band cnt group 4
```

;we have an applicable MAXSUBBANDS of 12 or 8

; group 4 is not applicable

```
move    x1,x:skfcnt4 ;zero the sub-band count for 4th group
move    #>8,x0      ;see if 3rd group gets zeroed also
cmp     x0,a      ;is applicable MAXSUBBANDS is 8
jgt     _sskfcrc_11 ;if MAXSUBBANDS = 12, continue
```

;we have an applicable MAXSUBBANDS of 3

; group 3 is not applicable

```
move    x1,x:skfcnt3 ;set count of sub-bands in 3rd crc
jmp     _sskfcrc_11
```

\_sskfcrc\_10

;set the 4th sub-band count based on applicable MAXSUBBANDS

254

BAD ORIGINAL

```

        move    y:<maxsubs,a          ;current MAXSUBBANDS
        move    =>12,x0              ;subtract 12 subs (for 1st 3 crcs)
        sub     x0,a                 ;set the sub-band count for 4th crc
        move    a,x:skfcrn4         ;set count of sub-bands in 4th crc

_sskfcrn_11
;do the scale factor checksum checks

        move    =SBndSKF,r0          ;addr of scale factors array
        move    =calskfck,r3        ;addr of calculated checksums
        move    =sbctls,r4          ;addr of table to control checksum

;indicate whether 2 channels n2 = 0: of mono (n2 = 1)

        move    =0,n2
        clr     #STEREO_vs_MONO,y:<stereo,_sskfcrn_30
        move    =1,n2

_sskfcrn_30
;calculate and test scale factor checksums

        do      #NUMSKFCKSUMS,_sskfcrn_40
        move    x:(r4)+,b            ;indicate starting sub-band number
        move    x:(r4)+,a            ;set number of sub-bands included

;see if the sub-band count is zero, and if so, skip the scale factor checksum

        tst     a                    ;check sub-band count for zero
        jne     _sskfcrn_33          ;if not zero, calc CRC checksum

;this sub-band group has no sub-bands, zero the CRC checksum

        clr     b                    ;set checksum to zero
        jmp     _sskfcrn_36          ;store zero checksum

_sskfcrn_33
;calculate the checksum (result is returned in b1)

        jsr     crcskf

_sskfcrn_36
;store the scale factor checksum

        move    b1,b                ;clean up checksum
        move    b,x:(r3)+           ;save scale factor checksum

_sskfcrn_40
;set up the checksums for storing in the previous frame

        move    =private,r2          ;to test type of prev frame
        move    =calskfck,r0        ;addr of calculated checksums
        move    =calskfck,r1        ;to store after alignment

;see if the previous frame was a split mono frame for bit duplication

```

```

;set #1,x:r2, _sskfcrc_50
move #>2,y1 ;store 2 formatted words
;not a split frame, concatenate pairs of 12-bit checksums into one 24-bit word
do y1,_sskfcrc_70
;get the next pair of checksums
move x:r2,-,a
move x:(r0)-,b
;left justify the 2nd checksum of the pair
do #12,_sskfcrc_50
asl b
_sskfcrc_50
;concatenate right justified 1st checksum with left justified 2nd checksum
move b1,a0
;shift left pair of checksums into a1
do #12,_sskfcrc_60
asl a
_sskfcrc_60
move a1,x:(r1)+ ;save the aligned pair
_sskfcrc_70
jmp _sskfcrc_140
_sskfcrc_80
;is a split frame, duplicate the 12-bit checksum bits into one 24-bit word
move #>4,y1 ;store 4 formatted words
do y1,_sskfcrc_140
;clear the target register and get the checksum
clr b x:(r0)-,a
;rotate right the bits from the right justified checksum and rotate them
;right in pairs into the target register
do #12,_sskfcrc_135
ror a ;bit to carry bit
;test the carry bit to see if zero or 1
jcs _sskfcrc_81 ;if carry bit set, so indicate
bclr #10,y:<not_appl ;carry bit = 1

```



```

        jmp      _sskfcrd_100                ;go duplicate the bit
_sskfcrd_90
        bset     #10,y:<not_appl              ;carry bit = 1
_sskfcrd_100
;now output 2 copies of the bit in the target register
        do       #2,_sskfcrd_130
        jset     #10,y:<not_appl,_sskfcrd_110 ;test if bit zero or 1
        andi     #SFE,ccr                    ;bit is a zero
        jmp      _sskfcrd_120                ;go insert the bit
_sskfcrd_110
        ori      #S01,ccr                    ;bit is a 1
_sskfcrd_120
;push the bit into the target register
        ror      b
_sskfcrd_130
        nop
_sskfcrd_135
;store the duplicated checksum for frame insertion
        move     bl,x:(r1)+
_sskfcrd_140
;now insert either the 2 or the 4 formatted words in the end of previous frame
;position to the scale factor checksums in the frame buffer
        move     x:skfcrdwd,r0                ;end of frame word address
        move     x:skfcrdcbt,a                ;bit offset to start skf crc's
        move     y:<outside,m0                ;circ buffer ctl
        move     #calskfck,r1                ;addr of formatted checksums
        move     y:(r0),b                    ;word from prev frame to start insert
        tst      a        #>24,x0            ;see if a right justify shift is needed
                                                ; & set bits per word
        jeq      _sskfcrd_150                ;no need to shift the 1st word
        sub      x0,a                        ;get bits to shift formatted word
        neg      a                          ;bits to shift the word to receive
;shift the formatted word to be ready to receive skf crc's
        do       a,_sskfcrd_150
        asr      b        x:skfcrdcbt,a      ;right justify bits
                                                ; & restore bit offset to start crc's
_sskfcrd_150
;for the number of formatted words, shift the bits into the previous frame

```

257

BAD-ORIGINAL



```

do      y1,_sskfcrc_160
move    x0,y1,-,b0      ;get formatted checksum word

;insert the 24 bits

do      #24,_sskfcrc_170

;see if full word shifted

cmp     x0,a      #>1,y0      ;test if 24 bits shifted
; & set bit shift incrementer
; if less than 24, shift bit in

;24 bits have been inserted, clear bits per word ctr and put word into prev frame

clr     a      ;zero bit shift counter
move    b1,y:(r0)-      ;formatted word to prev frame

_sskfcrc_160

;shift the bit into low order of word and count the bit inserted

asl     b      ;insert bit into b1
add     y0,a      ;increment word bit ctr

_sskfcrc_170
rop

_sskfcrc_180

;see if bits from frame buffer need to be formatted
;if an exact word fit, insert the newly formatted word

cmp     x0,a      a,y1      ;see if 24 bits in word
; & save bits inserted count
; if 24 bits, store the word

jeq     _sskfcrc_200

;we have to get the next word from the frame buffer and concatenate with the
;remaining bits of the scale factor checksums

move    #>24,a      ;to calc num bits to shift
sub     y1,a      ;determine bits to shift

;get the word at the next address

move    a,y0      ;save bits to left shift
move    y:(r0),a      ;get word from frame

;left shift the word from the frame same bits as shifted into b1

do      y1,_sskfcrc_190
asl     a      ;left justify in a1

_sskfcrc_190

;now concatenate the right justified checksum in b1 with left justified
; word from the frame

move    a1,b0      ;left justified frame word

```



```
;shift the remaining bits together for a fully formatted 34-bit word
    dc      y0,_sskfcrc_200
    asl     5
;store the 4th of the scale factor checksums in previous frame
    move    b1,y:cr01          ;word to frame buffer
    move    #1,m0              ;restore linear buffer ccl
;_sskfcrc_900
;indicate that a frame has been formatted
    cset    #2,x:private
    rts
```



```

opt    is

; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; XXXXXX: setbal.asm

title   'Set the bit allocations'

; New ISO frame format for stereo (2/7/91)
;
; This routine outputs the bit allocation bits.
;
; It is the ISO standard.
; sub-band 0 - 10 use 4 bits (11 * 4 = 44 bits)
; sub-band 11 - 22 use 3 bits (12 * 3 = 36 bits)
; sub-band 23 - 25 use 2 bits (4 * 2 = 8 bits)
;                                     total = 88 bits)

on entry
    r6 = current offset in output array
    y:<maxsubs = encoded sub-band range at:
        sampling rate, bit rate and whether MONO or 2 channels
    y:<sc = shift count
    y:opirtyp = full stereo, joint stereo or mono
    y:<stereo = type of framing flags used:
        bit 0 means stereo vs mono framing
            0 = stereo framing
            1 = mono framing
        bit 2 is to simply indicate that joint stereo applies
            0 = NOT joint stereo framing type
            1 = IS joint stereo framing type
        bit 3 is to indicate the full stereo upgrade by allocate rta
            if joint stereo applies
                0 = normal joint stereo allocation
                1 = FULL STEREO allocation
        bit 4 is to simply indicate the stereo intensity sub-band
            boundary has been reached if joint stereo applies
            0 = NO sub-bands still below intensity boundary
            1 = sub-bands above intensity boundary

    y:<sibound = for joint stereo sub-band intensity boundary
    x:crchits = accumulator of bits covered by CRC-16 routine
                (bit allocation bits are accumulated)

    r0 = address of left and right channels SubBandIndex array (x memory)

on exit
    a = destroyed
    b = destroyed
    y0 = destroyed
    y1 = destroyed
    r0 = destroyed
    r1 = destroyed
    r2 = destroyed
    r4 = destroyed
    n4 = destroyed

include 'def.asm'

section highmisc

```

```

xdef skfcb1_1
xdef skfcb1_2
xdef skfcb1_3
xdef skfcb1_4

org yhe:
stsetbal_yhe

; address of BAL's bit table as per Allowed table selected
skfcb1 ds 1

; These tables is the number of bits used by the scale factor in each sub-band
; High sampling rates with higher bit rate framing
skfcb1_1
dc 4 ;sub-band 0
dc 4 ;sub-band 1
dc 4 ;sub-band 2
dc 4 ;sub-band 3
dc 4 ;sub-band 4
dc 4 ;sub-band 5
dc 4 ;sub-band 6
dc 4 ;sub-band 7
dc 4 ;sub-band 8
dc 4 ;sub-band 9
dc 4 ;sub-band 10

dc 3 ;sub-band 11
dc 3 ;sub-band 12
dc 3 ;sub-band 13
dc 3 ;sub-band 14
dc 3 ;sub-band 15
dc 3 ;sub-band 16
dc 3 ;sub-band 17
dc 3 ;sub-band 18
dc 3 ;sub-band 19
dc 3 ;sub-band 20
dc 3 ;sub-band 21
dc 3 ;sub-band 22

dc 2 ;sub-band 23
dc 2 ;sub-band 24
dc 2 ;sub-band 25
dc 2 ;sub-band 26
;end table 3-B.2a
dc 2 ;sub-band 27
dc 2 ;sub-band 28
dc 2 ;sub-band 29
;end table 3-B.2b
dc 2 ;sub-band 30
dc 2 ;sub-band 31

; High sampling rates with lower bit rate framing
skfcb1_2
dc 4 ;sub-band 0
dc 4 ;sub-band 1

```

261

BAD ORIGINAL



```

dc      3      ;sub-band 2
dc      3      ;sub-band 3
dc      3      ;sub-band 4
dc      3      ;sub-band 5
dc      3      ;sub-band 6
dc      3      ;sub-band 7
;end table 3-3.20
dc      3      ;sub-band 8
dc      3      ;sub-band 9
dc      3      ;sub-band 10
dc      3      ;sub-band 11
;end table 3-3.21
dc      3      ;sub-band 12
dc      3      ;sub-band 13
dc      3      ;sub-band 14
dc      3      ;sub-band 15
dc      3      ;sub-band 16
dc      3      ;sub-band 17
dc      3      ;sub-band 18
dc      3      ;sub-band 19
dc      3      ;sub-band 20
dc      3      ;sub-band 21
dc      3      ;sub-band 22
dc      3      ;sub-band 23
dc      3      ;sub-band 24
dc      3      ;sub-band 25
dc      3      ;sub-band 26
dc      3      ;sub-band 27
dc      3      ;sub-band 28
dc      3      ;sub-band 29
dc      3      ;sub-band 30
dc      3      ;sub-band 31

```

; Low sampling rates

```

skfb1_3
dc      4      ;sub-band 0
dc      4      ;sub-band 1
dc      4      ;sub-band 2
dc      4      ;sub-band 3

dc      3      ;sub-band 4
dc      3      ;sub-band 5
dc      3      ;sub-band 6
dc      3      ;sub-band 7
dc      3      ;sub-band 8
dc      3      ;sub-band 9
dc      3      ;sub-band 10

dc      3      ;sub-band 11
dc      3      ;sub-band 12
dc      3      ;sub-band 13
dc      3      ;sub-band 14
dc      3      ;sub-band 15
dc      3      ;sub-band 16
dc      3      ;sub-band 17
dc      3      ;sub-band 18
dc      3      ;sub-band 19
dc      3      ;sub-band 20
dc      3      ;sub-band 21

```

262

BAD-ORIGINAL

```

        dc      2      ;sub-band 22
        dc      2      ;sub-band 23
        dc      2      ;sub-band 24
        dc      2      ;sub-band 25
        dc      2      ;sub-band 26
        dc      3      ;sub-band 27
        dc      2      ;sub-band 28
        dc      2      ;sub-band 29
end table 3-3.1
        dc      2      ;sub-band 30
        dc      2      ;sub-band 31

endsetbal_yhe
        endsec

        org      phe:

setbal
        move     y:skftbl,r1      ;get selected # of bits table address
        move     #NUMSUBBANDS,n0   ;access the right channel SBIndx
        bclr     #JOINT_at_SB_BOUND,y:<stereo      ;clear for initial sub-bands
        move     y:<sibound,r3      ;intensity stereo sub-band counter
        move     x:crcbits,r2      ;get CRC-16 bit counter

        dc      y:<maxsubs,_setb_40 ;output for applicable MAXSUBBANDS
        move     y:(r1)+,n4         ;get # of bits to use for this sub-band
        move     n4,n2              ;to accumulate CRC-16 bits
        move     x:(r0),y0          ;get left channel SubBandIndex(SubBand)
        jsr      setvalue           ;output the left channel value
        move     (r2)+n2            ;count bits covered by CRC-16 rtn

; if a mono type of frame, skip the right channel
        jset     #STEREO_vs_MONO,y:<stereo,_setb_30

; if not doing a joint stereo frame, handle the right channel
        jclr     #JOINT_FRAMING,y:<stereo,_setb_20

; if doing a joint stereo framing and frame upgraded to FULL stereo,
; handle the right channel
        jset     #JOINT_at_FULL,y:<stereo,_setb_20

; if joint stereo has reached the sub-band boundary, skip the right channel
        jset     #JOINT_at_SB_BOUND,y:<stereo,_setb_30

; check if the sub-band intensity boundary has been reached
        jsr      chkjoint

; if joint stereo has reached the sub-band boundary, skip the right channel
        jset     #JOINT_at_SB_BOUND,y:<stereo,_setb_30

_setb_20
        move     x:r0-n0,y0         ;get right channel SubBandIndex(SubBand)
        jsr      setvalue           ;output the right channel value
        move     r0-n2              ;count bits covered by CRC-16 rtn

```

```
_sacb_30  
    move    r0 -                ;next used sub-band  
_sacb_40  
    move    r1,x1crbbits        ;store updated CRC-16 bit counter  
    rts
```

264

BAD ORIGINAL

```

    opt    ic

;   © 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
;   XXXXXXsetanoda.asm
;
;   This routine outputs the ancillary data bytes to the output stream.
;
; on entry
;   r5 = current offset in output array
;   y:dataoptr = address in data byte input buffer to start from
;   y:bytecnt = count of bytes in input buffer not yet framed
;   y:maxbytes = max bytes per frame at given baud rate
;
; on exit
;   a = destroyed
;   b = destroyed
;   y0 = destroyed
;   y1 = destroyed
;   r0 = destroyed
;   r1 = destroyed
;   r4 = destroyed
;   n4 = destroyed
;
;   include 'def.asm'
;   include '..\common\ioequ.asm'
;   include 'box_ctl.asm'
;
;   section bytebuffer
;   xdef    databytes

    org    yhe:
stsetanoda_bytes

databytes    ds    DATABUFLEN    ;buffer for bytes received

endsetanoda_bytes
endsec

    section highmisc
    xdef    anctype
    xdef    baudrte
    xdef    dataiptr
    xdef    dataoptr
    xdef    bytecnt
    xdef    bytesfrm
    xdef    maxbytes
    xdef    ancbits
    xdef    padbits

    org    yhe:
stsetanoda_yhe

anctype      ds    1    ;type of count field after audio data:
;                   0 = 3 bit padded byte count
;                   1 = 3 bit data byte count
baudrte      ds    1    ;data baud rate code from switches
dataiptr     ds    1    ;ptr for next byte received
dataoptr     ds    1    ;ptr for next byte to insert into frame
bytecnt      ds    1    ;count of bytes yet to be output to frame

```

265

SAD ORIGINAL



```

bytesfrm      ds      1      ;count of bytes for output to current frame
maxbytes      ds      1      ;max bytes that can go at baud rate
ancbits       ds      1      ;bits in current frame for ancillary data
padbits       ds      1      ;unallocated audio bits to set pad byte count

endsetancdata_yhe
endsec

org phe:

setancdata

;if not ancillary data byte count,
;insert the count of pad bytes into the frame

move    #anctype,r4          ;to check for data byte count type
move    y:padbits,b          ;count of unallocated bits
jclr    #0,y:(r4),_ancd_00    ;if not data byte count, do pad byte cnt

;insert the count of ancillary data bytes rather than
;the CCS cdq standard count of pad bytes.

move    #BITSPERBYTE,n4      ;8 bits for byte count
move    y:bytesfrm,y0        ;count of ancillary data bytes
jmp     _ancd_05              ;insert the byte count

_ancd_00

;normal CCS cdq's encode the byte count of unallocated MUSICAM bits
;divide number of unallocated bits by 8 (bits per byte) to get
;truncated count of total bytes padded with 0

lsr     b                    #BITSPERBYTE,n4 ;divide by 2
; & set number of bits for pad count
lsr     b                    ;divide by 2 again (==> by 4)
lsr     b                    #0,x0          ;divide by 2 again (==> by 8)
; & get set to zero count

tst     b                    ;should never be negative
clt     x0,b                  ;if negative, set to zero
move    b1,y0                ;set up to insert pad count

_ancd_05

;encode the padded byte count or ancillary data byte count

jsr     setvalue              ;insert byte count for decoder

move    y:bytesfrm,b          ;if count of data bytes is zero
tst     b                    ;test if no bytes this frame
jeq     _ancd_100             ;no data bytes to insert

;now insert the bytes into current frame.

move    y:dataoptr,r5         ;address of next byte to be output
move    #DATABUFLEN-1,r5     ;circular buffer
move    #BITSPERBYTE,n4      ;number of bits to insert in the frame
do      b,_ancd_10           ;output the number of bytes
move    y:r5+7,y0            ;word with the byte to insert
jsr     setvalue              ;format the byte in the frame

```



```

        nop

;_ancd_10
;temporarily disable data received interrupt to decrement unframed byte count
        bclr    #M_RIE,x:<<M_SCR

;while waiting for disable interrupt to take effect:

        move    r5,y:datacptr          ;save addr of next byte for next frame
        move    #-1,m5                 ;uncircular buffer
        move    y:bytesfrm,y0          ;count of data bytes just framed

;interrupt should be cleared by now to safely get byte count maintained by
;interrupt routine

        move    y:bytecnt,a            ;get latest byte count of unframed bytes
        sub     y0,a    #0,y0          ;subtract count of bytes just framed
        ; & get set to zero count
        blt     y0,a                   ;if negative, zero count
        tst     a                      ;make sure we're not negative
        jge     _ancd_20               ;if 0 or more, continue
        clr     a                      ;reset to zero (just a precaution)

;_ancd_20
        move    a,y:bytecnt            ;save new unsent byte count

;turn the receive byte interrupt back on
        bset    #M_RIE,x:<<M_SCR      ;reable receive interrupt

;_ancd_100
;pad 0 bits to the end of the audio portion of the frame

        move    y:audendpos,r0         ;get bit count to end of MUSICAM frame

;set flag for reed solomon (if reed solomon, skip the frame flush)

        move    #reedsolomon,r4        ;addr of the flag
        jsr     flushframe              ;pad frame with zeroes to MUSICAM end

        tst     a                      ;see if an overshoot ?????
        jlt     _ANCD_HELP              ;OVERSHOOT ERROR!!!!!!
        jmp     _ancd_150               ;OK, see if any client trailing bits

;;pad 0 bits to the end of the audio portion of the frame
;;
;;        move    #0,y0                 ;init with zeros to pad last word
;;        move    y:audendw,x1          ;address of end of audio portion
;;        move    r6,b                  ;next o/p addr of current frame
;;        cmp     x1,b    #>24,a       ;if addresses eq, handle last few bits
;;        ; & set up for the next test
;;        jeq     _ancd_130             ;we're at the last word of audio
;;
;;output last partially formatted data word before zero fill remainder of frame
;;
;;        move    y:<sc,x0              ;get number of bits in last word

```

267

```

;;      sub      x0,a      =>24,x0      ;get number of bits left
;;      cmp      x0,a      =0,x0      ;24 - number of bits left
;;      jeq      _ancd_110      ;not partially formatted (y:sc == 0)
;;
;;      move     y:<curwd,b      ;get current output word
;;      rep      a      ;output the necessary # of bits
;;      lsl      b
;;
;;      move     b1,y:(r6)-      ;save in the output
;;      move     x0,y:<sc      ;zero the current bit offset
;;
;;_ancd_110
;;
;;      clr      a      ;output zero for remainder of frame
;;
;;_ancd_120
;;
;;;see if the last word of the audio portion of frame is to be output next
;;
;;      move     r6,b      ;next o/p address of current frame
;;      cmp      x1,b      ;see if last word next
;;      jeq      _ancd_130      ;last word, chk for any remaining bits
;;      move     a1,y:(r6)-      ;output frame word and increment addr
;;      jmp      _ancd_120      ;continue to flush the buffer
;;
;;_ancd_130
;;
;;;handle the last word of the frame
;;
;;      move     y:audendb,b      ;bit offset signaling end of audio
;;      move     y:<sc,y1      ;get current formatted word offset
;;      sub      y1,b      ;sub to get # bits remaining
;;      tst      b      ;test if any zero bits to output
;;      jeq      _ancd_150      ;if none, we're done
;;      jgt      _ancd_140      ;OK, output value
;;
;;_ANCD_HELP
;;
;;ERROR!!! this case should not occur
;;
;;      ON_BITALLCC_LED_CD      ;!!! error we've overshoot
;;
;;;!!!debug: dump the frame in question (pull of the ';' from next line)
;;
;;      jsr      dumpdata
;;
;;      jmp      _ancd_150
;;
;;_ancd_140
;;      move     b,r4      ;number of bits to output
;;      jsr      setvalue      ;pad word with zeroes as needed
;;
;;_ancd_150
;;
;;insert any client trailing bits
;;
;;      INSERT_CLIENT_TRAILING_BITS_CD
;;
;;      rts

```

268

BAD-ORIGINAL

```

opt f0
; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
; UXC003B\scirec.asm

title 'SCI receive ancillary data interrupt handler'

include 'def.asm'
include '...\common\ioequ.asm'

; these save variables for exclusive use by the scirec interrupt handlers only

section highmisc
xdef    scirecR7Save
xdef    scirecN7Save
xdef    scirecM7Save

org     xhe:
stscirec_xhe

scirecR7Save    ds    1
scirecN7Save    ds    1
scirecM7Save    ds    1

endscirec_xhe
endsec

; SCI xcode receive ancillary data interrupt

org     pli:

scirec
move     r7,x:scirecR7Save
move     m7,x:scirecM7Save

move     y:dataiptr,r7           ;get input data byte buffer pointer
move     #DATABUFLEN-1,m7       ;circular buffer
nop
movep    x:<<M_SRXL,y:(r7)+      ;get the byte and store in buffer
move     r7,y:dataiptr          ;update input data byte buffer pointer
move     y:bytecnt,r7           ;increment the data byte counter
move     #-1,m7                 ;no circular buffer ctl for count
nop
move     (r7)+                   ;increment
move     r7,y:bytecnt           ;save the new byte count

move     x:scirecM7Save,m7
move     x:scirecR7Save,r7

_sci_90
rti

; SCI xcode receive ancillary data interrupt exception

scirece
move     r7,x:scirecR7Save

movep    x:<<M_SSR,r7           ;clear the exception
movep    x:<<M_SRXL,r7         ;eat the byte

```

269



```
move    x:scirecR7Save,r7  
rti
```

270

BAD-ORIGINAL



```

;-----
;
; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; UKCODE\quantize.mac
;
QUANTIZE macro
;
; This routine is used to quantize the data.
; The resulting data is right justified in the result register.
;
; 1st test to see if we are doing a Joint stereo quantize and if so,
; do the joint quantize routine and return the result from that routine;

    jclr    =JOINT_at_SB_BOUND,y:<stereo,_quant_20

    move    y0,a                ;get value to test register
    move    y:MaxiFact,y0      ;get the Maxi scale factor
    tst     a                   #lshftb1,r4    ;see if dividend is negative
    jlt     _jquan_10          ;it is

; - dividend and - divisor

    move    y:(r4+n4),y1
    and     #Sfe,ccr           ;clear the carry bit
    rep     n4                 ;value/scalefactor
    div     y0,a
    div     y0,a               ;one more div
    div     y0,a               ;one more div

    move    a0,y0
    mpy     y0,y1,a            #qstb1,r4      ;get result to a reg
    jmp     _jquan_20         ;left justify

; - dividend and - divisor

_jquan_10
    neg     a                   y:(r4+n4),y1    ;make +
    and     #Sfe,ccr           ;clear the carry bit
    rep     n4                 ;value/scalefactor
    div     y0,a
    div     y0,a               ;one more div
    div     y0,a               ;one more div

    move    a0,y0
    mpy     -y0,y1,a           #qstb1,r4      ;get result to a reg
                                         ;left justify

_jquan_20
    move    a0,a
    tfr     x1,a               a,y0
    mac     x0,y0,a            y:(r4+n4),y1    ;form quantized result
    asr     a                   y:<bitscnt,r4    ;divide by 2
                                         ; & get bits used so far

    move    a,y0
    mpy     y1,y0,a            y:<sc,y1        ;right justify the bits
                                         ; & = of bits left in curr word

; done with joint quantizing, go to the end of the macro

    jmp     _quant_300

```

271

BAD ORIGINAL



\_quant\_90

; This routine assumes that it must multiply two numbers together.  
 ; One number is called P and the other number is called Q.  
 ; P is unsigned and consists of an integer part (24 bits) and a  
 ; fractional part (24 bits).  
 ; Q is a signed fractional number (24 bits).

; P is of the form P1.P0 and Q is of the form .Q0 .  
 ; The produce of P \* Q is always less than 1.

; To perform the multiplication,

```

      P1.P0
      *   .Q0
      -----
      .P0Q0
      P1.Q0
  
```

; To do this in the dsp, assume the following register usage

```

;
; P1 = y1
; P0 = y1
; Q0 = y0
  
```

; the result (to 24 bits) is in a (as a signed value)

```

;3/24/94  move    x:(r5+n5),y1      ;get P0
          move    a,y0             ;get Q0 in right register
          mpy     y1,y0,a          ;P1 * Q0
          asr     a                y:(r5+n5),y1 ;rslt will always be in a0
                                           ;adjust for integer * fractional
          move    a0,a            ;move to right position
                                           ; in accumulator
          macr    y1,y0,a         #qstbl,r4 ;P0 * Q0
          tfr     x1,a            a,y0
          mac     x0,y0,a         y:(r4+n4),y1 ;form quantized result
          asr     a                y:<bitscnt,r4 ;divide by 2
                                           ; & get bits used so far
          move    a,y0
          mpy     y1,y0,a         y:<sc,y1 ;right justify the bits
                                           ; & # of bits left in curr word
  
```

\_quant\_900

```

endm
list
  
```

272

BAD ORIGINAL

```

        opt      fo,mex
;
; 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; UXCODO\qcalcglo.asm uses lower.asm and upper.asm - Larry values!
;
        title    'Calculate Global Masking Threshold'
;
; This routine is used to calculate the global masking threshold.
;
; on entry
;       r4 = address of masker structure (l memory)
;       r1 = address of GlobalMaskingThreshold (in slb's) (x memory)
;       x:<nmasker = number of maskers
;
; on exit
;       a = destroyed
;       b = destroyed
;       x0 = destroyed
;       x1 = destroyed
;       y0 = destroyed
;       y1 = destroyed
;       r2 = destroyed (pfmap)
;       r3 = destroyed (lmskr)
;       r4 = destroyed (rmskr)
;       r5 = destroyed (b_i)
;       r6 = destroyed
;       n1 = destroyed (thrsld - current index into Threshld
;       n2 = destroyed (mskrnum)
;       n4 = destroyed
;       n5 = destroyed (k)
;
;       include 'def.asm'
;       include '..\xlpsycho\lower.asm'
;       include '..\xlpsycho\upper.asm'
;       include '..\uxcode\dbadd.mac'
;
        org      pli:
QCalcGlo
; note: r4 is now free and could be used for Threshld

        move     #0,n2                      ;set to working on first mskr
        move     y:thresslb,n1              ;start of threshold array (SLB)
; Find first masker which is not deleted.

        move     #>DELETEDMSKR,x0          ;deleted type
        move     #MASKERSTYPE,n4           ;offset to type

        move     x:<nmasker,b               ;get number of maskers
        tst      b,y:pfmap,r2              ;and check for non zero
; & pfmap = fmap

        jeq      <_calc_10
        do       b,_calc_10
        move     x:r4+n4/,a                 ;get type
        cmp      x0,a,#MASKERSSIZE,n4      ;check if deleted
        jeq      <_calc_05

```

273

BAD ORIGINAL



```

        enddo
        jmp      <_calc_10

; found a non-deleted masker

_calc_05
        move     (r4)+n4
        move     #MASKERSTYPE,n4
        nop

_calc_10
        do       y:<nmskfreqs,_calc_00

        move     r1,r5
        move     y:(r2)+,n5
        move     #MASKERSBFREQ,n4
        move     x:(r5+n5),a
        move     y:b_i,r5
        move     a,x:(r1)

        move     y:(r5+n5),y0
        move     y:(r4+n4),a
        cmp      y0,a      #MASKERSTYPE,n4
        jgt      <_calc_30

        move     #MASKERSSIZE,n4
        move     r4,r3
        move     (r4)+n4

; get address of next quiet pwr
; k = *pfmap++
; offset to BFreq
; get the quiet power in SLB's
; get base address of b_i table
; save as power in SLB's

; BFreq = b_i
; rmskr->BFreq
; rmskr->BFreq = BFreq

; size of the structure
; rmskr = rmskr
; ++rmskr

; Find next masker which is not deleted.

        move     #>DELETEDMSKR,x0
        move     x:<nmasker,b
        tst      b      #MASKERSTYPE,n4
        jeq      <_calc_20

        do       b,_calc_20
        move     x:(r4+n4),a
        cmp      x0,a      #MASKERSSIZE,n4
        jeq      <_calc_25

        enddo
        jmp      <_calc_20

; found a non-deleted masker

_calc_25
        move     (r4)+n4
        move     #MASKERSTYPE,n4
        nop

_calc_20
        move     #MASKERSTYPE,n4
        move     #>1,n2

; set to not the first masker

_calc_30
        move     x:(r4+n4),a
        move     #>ENDMSKR,x0
        cmp      x0,a      #MASKERSBFREQ,n4
        jeq      <_calc_40

        move     y:(r4+n4),b

; rmskr->Type
; end type
; if at end don't process right
; rmskr->BFreq

```

274

BAD-ORIGINAL



```

sub     y0,b     #.09375,x0           ;sdbark = lmskr->BFreq - BFreq
cmp     x0,b     #MASKERSPWDRDB,n4    ;sdbark = .09375
jgt     <_calc_40                     ;check range

move     #.03125,x1
move     x:(r4+n4),y1                 ;lmskr->PowerDB

LOWER_SLOPE

move     y:(r4+n4),a
sub     x1,a     x:(r1),x0           ;lmskr->PowerDB
                                           ;form masking skirt
                                           ; and get GlobalMasking Threshld
DBADD
move     b,x:(r1)

move     y:(r5+n5),y0                 ;BFreq = b_i

_calc_40
move     n2,a
tst     a     #MASKERSBFREQ,n3
jeq     <_calc_50

move     y:(r3+n3),b
sub     y0,b     #.25,x0           ;lmskr->BFreq
neg     b     #MASKERSPWDRDB,n3    ;lmskr->BFreq - BFreq
cmp     x0,b     #.03125,x1        ;BFreq - lmskr->BFreq
jgt     <_calc_50

move     x:(r3+n3),y1                 ;get the ->lmskr->PowerDb

UPPER_SLOPE

move     y:(r3+n3),a
sub     x1,a     x:(r1),x0           ;lmskr->ReducedPowerDb
                                           ;form masking skirt
                                           ; & get GlobalMaskingThreshld
DBADD
move     b,x:(r1)

_calc_50
move     (r1)+

_calc_90
rts

```

```

                                opt      cex
;
;  © 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;  UXCODE\findtona.asm
;
;  title    'find tonals'
;
;  This function is used to find the tonals. Once the tonal is found, it is
;  replaced by a single power value which is the sum of 3 points.
;
;  on entry
;      r1 = address of the power array (1 memory)
;      r2 = address of the tonal structure (1 memory)
;      r4 = address of the range table (y memory)
;
;  on exit
;      r3 = # of tonals found
;
;      a = destroyed
;      b = destroyed
;      x0 = destroyed
;      x1 = destroyed
;      y0 = destroyed
;      y1 = destroyed
;      r1 = destroyed
;      r2 = destroyed
;      r5 = destroyed
;      n1 = destroyed
;      n2 = destroyed
;      n4 = destroyed
;
;  include 'def.asm'
;
;  org      phe:
;
findtona
;
;      move      r1,r5                      ;save starting address
;
;      First compute the ending address
;
;      move      #>509,y1
;      move      #>320,y1
;      move      r1,a
;      add       y1,a
;      move      a,y1                      ;save ending address for later
;
;      move      (r1)+                      ;start at power + 2
;      move      r1+
;      move      #0,r3                      ;ntonals = 0
;      move      #-1,n1
;
;      This is the big loop where we look for tonals
;
;  _find_20
;
;      look for a local maximum
;
;      move      1,r11,b                    ;power

```

276

BAD-ORIGINAL

```

move    l:(r1-n1),a          ;pow[i-1]
cmp     a,b    #1,n1          ;pow[i] - pow[i-1]
jle     _find_39
move    l:(r1+n1),a          ;pow[i+1]
cmp     a,b    #1,n1          ;pow[i+1] - pow[i]
jle     _find_39

; scale pow[i] down by 7.2 dB (should be 7.0 dB)

asr     b                      ;get 3/16 of power
asr     b
tfr     b,a                    ;move entire register
asr     b
asr     b
sub     b,a                    ;power is down 7.2 dB

; scale pow[i] down by 6.0 dB (ISO says 7.0 dB)

asr     b                      ;get 1/4 of power
asr     b
tfr     b,a                    ;power is down by 6.0 dB

; Now search on each side to see if a tonal.

; first determine search range.

move    r1,b                  ;get the current index of pow
move    r5,x1                  ;get starting position
sub     x1,b    #2,y0          ;compute distance into array
move    b1,x1                  ;move to right register
mpy     x1,y0,b    #>1,x0      ;shift right 6 bits
move    b1,n4                  ;get offset into range table
move    r1,n2                  ;save starting r1 value
move    y:(r4+n4),b            ;get range
sub     x0,b    (r1)-          ;range - 1
move    b,x0                   ;save range - 1

; search the lower side
; must back off two from center address. one backoff was done above.

move    (r1)-                  ;set r1 to starting value

move    l:(r1)-,b              ;get first power value
do      x0,_find_40            ;search range

; cmp     b,a                    ;3/16 * pow[i] - pow[i-j]
; cmp     b,a                    ;1/4 * pow[i] - pow[i-j]
; jge     _find_30              ;so far so good

enddo
move    n2,r1                  ;restore r1
jmp     _find_39

_find_30
; _find_40
move    l:(r1)-,b              ;get next power value

; now search the upper side

move    n2,r1                  ;restore r1

```



```

nop
move    r1, -           ;set r1 to starting value
move    r1, -

move    1:(r1),b         ;get first power value
do      x0,_find_42       ;search range

cmp     b,a              ;3/16 * pow[i] - pow[i-j]
jge     _find_32         ;so far so good

enddo

move    n2,r1            ;restore r1
jmp     _find_39

_find_32
move    1:(r1),b         ;get next power value
_find_42

move    n2,r1            ;restore r1

now we save the bin number in the tonal structure

move    #TONALSBIN,n2    ;get bin offset
move    #-1,n1           ;set index
move    x1,x:(r2+n2)     ;save the fft bin number

we found a local maximum and it was a tonal
add power of 3 highest points

move    1:(r1),b         ;pow[i]
move    1:(r1+n1),a       ;pow[i-1]
add     b,a              ;pow[i+1]+pow[i]
move    #TONALSPWRDB,n2   ;get offset to power
move    1:(r1+n1),b       ;pow[i-1]
add     b,a              ;pow[i+1]+pow[i]+pow[i-1]
move    a,1:(r2+n2)      ;save in tonal array

```

.....

Now advance the r1 position to next possible position.  
The next possible position is the current position + range+1.  
We only advance it by range since the +1 is done at the bottom  
of the loop.

```

move    r1,x0           ;get range
add     x0,b            ;r1 + range
move    b1,r1

```

.....

10-8-91

Now advance the r1 position to next possible position.  
The next possible position is the current position - 2 - 1.  
We only advance it by 2 since the -1 is done at the bottom  
of the loop.  
This advancement is less than the old method because the old  
method skipped over tonals which were higher and the the skipped  
tonal was then considered as noise and generated a higher  
masking threshold. This caused less bits to be allocated to  
the sub-band than there should have been.  
Remember that the energy in a tonal is the sum of the power in

278

BAD-ORIGINAL

the highest point and the left and right hand points-  
around the highest point.

move r11-  
move r11-

We come here when we have finished processing a tonal and put it in  
the tonal structure.

move #TONALSSIZE,n2 ;get size of tonal structure  
move r31- ;tonals++  
move r21-n2 ;advance to next entry

\_find\_33

move r11- ;start looking at next point

move r1,a ;get current count  
cmp y1,a #-1,n1 ;get maximum count  
jle \_find\_00

rts

end



```

opt 3c

; 1991. Copyright Corporate Computer Systems, Inc. All rights
; reserved.
; UKCODE\bitsallo.asm

title    'Initialize bit output'

; routines:
; setframelen: Sampling Rate 44100 & sampling at 33000 for 399
; kbs:
; This routine handles the test for whether frames
; need to be padded and set the working length (y:bitsfrm)
; for the next frame as it performs the necessary ISO
; formula
; updates for the next frame. A padded frame length is
; y:frmbits plus 9 bits;
; Other Sampling Rates require no padding. In this case
; the working frame bit length (y:bitsfrm) is set equal
; to y:frmbits.

include  'def.asm'
include  'box_ctl.asm'

org phe:

setframelen

; set the working frame length in bits for the current frame to be
; coded:
; if the frame requires no padding (most cases), y:bitsfrm =
; y:frmbits
; determine if the frame is to be padded:
; get frame's unpadded bit count
; get current REST value (if not negative, no padding)
; initialize as no padding in this frame (set code for frame
; header)
; get the DIFF value at the sampling rate and framing bit rate

move y:frmbits,b      ;unpadded frame bit length
move y:paarest,a      ;REST after last frame
move #0,y1             ;indicate no padding
test a,y:usediff,x0    ;see if padding needed.
jge _padd_00          ;if not neg, no padding

;this frame is padded, add the number of bits as per ISO to normal
; frame length
; and set the indication for the frame header that the frame is
; padded

```

; add the sampling frequency to REST as part of calculation for the next frame

```

move 4>PAD_SLOT,x1      ;padded bits added to frame
add x1,b 7:padrate,y0    ;add to unpadded frame bit length
                        ; & get the sampling rate value
add y0,a #>1,y1         ;add sampling rate to REST
                        ;set padded indication for frame header

```

\_padd\_00

;decrement the REST variable by the DIFF value for the next frame

```

sub x0,a                ;sub the DIFF value from REST
move a,y:padrest        ;save update REST value for next
frame

```

;indicate if padded or not as determined above (for frame header)  
;and set the frame in bit length

```

move y1,y:usediff       ;indicate if padded or not
move b,y:<bitsfrm        ;set bits in the frame
rts

```

;bitpool()

; This subroutine determines the number of bits available based  
; on the output bit rate and the type of framing

\*\*\*\*\*  
\*\*\*\*\*

;The table below is based on a Sampling Rate at 48,000 /sec and  
shows

;the breakdown of bit counts based on bit rate o/p and choice of  
frame type

			Mono		Full		Joint Stereo				
							4-bound	8-bound	12-bound		
kb frame					Stereo						
16-bound											
rate	bits	fix	avail		fix	avail	fix	avail	fix	avail	
avail	fix	avail									
384	9216	136	9080	224	8992	152	9064	168	9048	183	9033
195	9021										
256	6144		6008		5920		5992		5976		5961
	5949										
192	4608		4472		4384		4456		4440		4425
	4413										
128	3072		2936		2848		2920		2904		2889
	2877										
112	2688		2552		2464		2536		2520		2505
	2493										
96	2304		2168		2080		2152		2136		2121
	2109										



```

; 64 1536      1400      1312      1384      1368      1353
;      1341
; 65 1344      136      1268      124      1120      152      1192      168      1176      133      1161
; 135 1149
; .....
; .....
;
; y:<sibound = for joint stereo this is the sub-band boundary
;              below which sub-bands are full stereo
;              otherwise,
;                  only one channel (the left) is accounted for
; y:<stereo = flags:
;             bit 0 means stereo vs mono framing
;                 0 = stereo framing
;                 1 = mono framing
;             bit 2 is to simply indicate that joint stereo applies
;                 0 = NOT joint stereo framing type
;                 1 = IS joint stereo framing type
;             bit 3 is to indicate the full stereo upgrade by
allocate rtn
;             if joint stereo applies
;                 0 = normal joint stereo allocation
;                 1 = FULL STEREO allocation
;             bit 4 is to simply indicate the stereo intensity
sub-band
;             boundary has been reached if joint stereo applies
;                 0 = NO sub-bands still below
intensity boundary
;                 1 = sub-bands above intensity
boundary
;             bit 11 does dual line transmission apply requiring
that a
;             block sequence number be appended to the coded
frame
;                 0 = dual line block sequence does NOT
apply
;                 1 = dual line block sequence
numbering APPLIES
;             bit 18 indicates whether or the crc checksum applies
;                 0 = NO do not account for checksum
;                 1 = YES do account for checksum
;
; y:<maxsubs = maximum sub-bands at sampling rate, bit rate & 1
vs 2 chans
; y:<bitsfrm = the total number of bits in a frame at the
specified
;             bit rate if applicable, padded frame bits were
added
;             to y:<frmbits)
; these are used to determine if the frame requires a pad of 3
bits
; y:padrate = sample rate value
; y:padrest = updated REST value in ISO calculation
; y:usediff = DIFF value in ISO calculation after determining

```



```

; whether padding is necessary, this variable is changed:
; 0 = NOT a padded frame
; 1 = frame was padded

; on exit:
; x0 destroyed = returned number of required (fixed) bits
; x1 destroyed = returned number of bits available for bit
allocation

; a destroyed
; b destroyed
; r0 destroyed
; r1 destroyed
; r3 destroyed
; r4 destroyed

section lowmisc
xdef sc,curwd,bitsfrm,bitscnt

org yli:
stbitsallo_yli

sc ds 1 ;shift count
curwd ds 1 ;current word
bitsfrm ds 1 ;bit length of the current frame
bitscnt ds 1 ;count bits inserted in frame

endbitsallo_yli
endsec

org phe:

bitpool

;Select the proper Allowed table:
; ISO:
; 1. for low sampling rates (24 or 16 K),
; set ISO Extention Allowed table (Allowed_3)
; 2. for high sampling rates (48, 44.1 or 32 K):
; a. based on MAXSUBBANDS less than 27,
; set ISO lower bit rate Allowed table (Allowed_2)
; b. else,
; set ISO higher bit rate Allowed table (Allowed_1)
; CCS:
; set ISO higher bit rate Allowed table (Allowed_1)

;low sampling rate:
; test the frame header ID bit if 0, it's a low sampling rate
frame

move #smplidbit,r0 ;addr of frame header ID bit 0 =
low) nop ;1 = high)

```



```

;set #0,y: r0, _bitp_000_A ;if high rate, select Allowed
table

move #Allowed_3,r0 ;addr of low sampling allowed table
move #skftbl_3,r1 ;addr of the BAL bits table
move #>15,x1 ;maximum position Allowed_3 table
jmp _bitp_010_A ;go to store Allowed table address

_bitp_000_A

;high sampling rate:
; set the proper Allowed table address based on working MAXSUBBANDS
;y:<maxsubs)
; if less than 27, used table 2

move y:<maxsubs,x0 ;get current MAXSUBBANDS
move #>27,a ;to see which of 2 tables applies
move #>17,x1 ;maximum position Allowed_1 table
move #skftbl_1,r1 ;addr of the BAL bits table
cmp x0,a #Allowed_1,r0 ;see if need the low bit rate table
; & set up as Allowed_1 table
jle _bitp_010_A ;Allowed_1 table applies

;select the lower bit rate Allowed table

move #Allowed_2,r0
move #skftbl_2,r1 ;addr of the BAL bits table
move #>16,x1 ;maximum position Allowed_2 table

_bitp_010_A

;set the address of the selected Allowed table
;set the address of the selected BAL's bit table
;set the maximum position code

move r0,y:<AllwAdd
move r1,y:skftbl
move x1,y:MaxPos

;determine the bits required for ancillary data (taken from audio
pit pool):
; start with bits required to store the padded data byte count in
frame

move #anctype,r4 ;to see if data byte count applies
move #>BITSFORPADDING,b ;bits in the padded byte count

;if data byte count applies, change padded bits byte count 3 bits
; to count (3 bits) of ancillary data bytes encoded in the frame

jclr #0,y:(r4),_bitp_00 ;if not data byte, proceed
move #>BITSFERBYTE,b ;size of the ancillary data byte
count

```



```

_bitp_00
    move y:maxbytes,y1      ;get max bytes at baud rate
    move y:bytecnt,a        ;get current count of bytes received
    cmp y1,a #>BITSPERBYTE,x1 ;see max versus current count
                                ; & set multiplier
    jge _bitp_05            ;if more than max, can only send max
    move a,y1               ;less than max, send all received

_bitp_05
    ;multiply the bytecount for bits per byte

    mpy x1,y1,a             ;to get the required bit
    asr a y1,y:bytesfrm     ;shift integer result
                                ; & set byte count for framing
    move a0,a
    add a,b                 ;add bits to bits in byte count field

    ;!!!!test
    ;!!!!tst move y:<bitsfrm,b ;!!!!test: get total frame bits
    ;!!!!tst lsr b #0,x1      ;!!!!test: take half for ancillary
data
    ;!!!!tst lsr b #0,x1      ;!!!!test: take quarter for ancillary
data
    ;!!!!tst move x1,y:bytesfrm ;!!!!test: zero byte count for frame
    ;!!!!test
        move b,y:ancbits     ;set ancillary data bit count

    ;set the number of fixed bits used, and the number of available
bits for audio

    clr a #0,x1             ;0 a as accum, zero CRC checksum bit
cnt

    ;set the fixed bits for the audio frame

    move #>NSYNC,x0          ;number of SYNC bits
    add x0,a #>NSYST,x0      ;plus number of bits in frame system
hdr
    add x0,a y:skftbl,r0     ;get base of used bits table

    jclr #PROTECT,y:<stereo,_bitp_35 ;skip checksum bits if no
protect
    move #>NCRCBITS,x1       ;add applicable bits for the checksum

_bitp_35
    add x1,a                 ;add checksum protection, if any

    ;in case of Joint stereo, set the intensity sub-band boundary value

    move y:<sibound,r3

    ;accumulate the bit allocation bits for standard number of

```

```

sub-bands
; included in the frame for the left and right if applicable
    do y:<maxsubs,_bitp_50
; always accumulate for the left channel
    move y:r0!-,x1
    add x1,a
; if doing one channel only, skip the right channel
    jset #STEREO_vs_MONO,y:<stereo,_bitp_40
; if NOT doing joint stereo framing or framing at FULL stereo,
; add for the right channel
    jclr #JOINT_FRAMING,y:<stereo,_bitp_30
    jset #JOINT_at_FULL,y:<stereo,_bitp_30
; if doing Joint and we have reached the intensity sub-band
boundary,
; skip the right channel
    jset #JOINT_at_SB_BOUND,y:<stereo,_bitp_40
; if doing Joint check to see if we have reached the intensity
sub-band boundary,
; and if we just did, skip the right channel
    jsr chkjoint
    jset #JOINT_at_SB_BOUND,y:<stereo,_bitp_40
_bitp_30
; stereo, add for the right channel
    add x1,a
_bitp_40
    nop
_bitp_50
    move a,x0 ; return fixed bits
    move y:<bitsfrm,b ; total size of frame in bits
    move b,y:frmendpos ; bits to end of total frame
    move b,y:audendpos ; bits to end of MUSICAM frame
    move b,y:bsnendpos ; bits to end location for block seq
num
    move b,y:reedendpos ; bits to end total frame reed
solomon;
; if doing a split mode transmission, subtract the bit for the block
sequence

```

```

        jclr #SPLIT_MODE,y:<stereo,_bitp_70
        move =>BLOCK_SEQ_NUM_BITS,x1
        sub x1,b
        move b,y:bsrendpos          ;bits to end location for block seq
num
_bitp_60
; if formatting a MONO split frame, divide the formatted frame bits
in half
        jclr #SPLIT_MONO_FRAME,y:<stereo,_bitp_70 ;if NOT, continue
;***** (start) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****
;***** (start) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****
; divide the formatted frame bits in half
        lsr b
        move b1,b
;***** (end) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****
;***** (end) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****
_bitp_70
; if reed solomon frames, subtract the reserved bits from the bit
pool
        move #reedsolomon,r4          ;addr of the flag
        nop
        jclr #0,y:(r4),_bitp_75 ;if no reed solomon bits, continue
        move y:trailbits,y1          ;get bits required for this frame
        sub y1,b                      ;reduce the bit pool
_bitp_75
; save end bit position of the MUSICAM frame
; plus any bits required for client trailing bits and scale
factor crc's
        move b,y:frmendpos          ;bits to the end encoded frame
; subtract any trailing bits required by the client
; and, if applicable, subtract bits for the next frame's scale
factor checksums
        move #private,r0              ;to see if skf crc's apply
        move =>CLIENT_TRAILING_BITS,y0 ;get count of client reserved
bits

```

```

sub y0,b =>NSKFORCBITS*NUMSKFCKSUMS,y0 ;sub client bits
; & set count of skf checksum bits
clr #0,x:r0,_bitp_80 ;if not appl, do not sub skf crc bits
sub y0,b ;sub bits for skf checksum bits

_bitp_80

;set bit count to the end of the MUSICAM frame

move b,y:audendpos ;end MUSICAM up to client bits & skf
crc

;subtract the bits required for ancillary data

move y:ancbits,y1 ;get count of ancillary data bits
sub y1,b ;less the ancillary data bits

;subtract the accumulated frame fixed bits

sub a,b ;total bits - fixed bits

;this leaves the bits available for allocation

move b,x1 ;return number of audio data bits avail.

;done in all cases with end bit positions set

rts

;bitsallo()
; This subroutine starts the bit allocation of values into the
; frame buffer values are inserted by setvalue() and by
; bitfree() below

; on entry
; r5 = address of the output buffer
; m6 = circular buffer control for OutData (2 frames 2*frame
wds)

; on exit
; y:sc = 0
; y:curwd = initialized (0) 1st word in frame buffer
; r6 = address of the output buffer
; m6 = circular buffer control for OutData (2 frames 2*frame
wds)
;
; a = destroyed

bitsallo
clr a
move a,y:<sc ;initialize the shift count
move a,y:<curwd ;initialize curwd 1st bit in op
frame)
move a,y:<bitscnt ;start the bit counter of framed bits

```

```

        rts

;bitsfree:
;   This routine flushes the last bits to the output buffer
;
; on entry
;   r6 = address of next word the output frame buffer (y memory)
;   y<stereo bit 11 does dual line transmission apply requiring
that a
;           block sequence number be appended to the coded
frame
;           0 = dual line block sequence does NOT
apply
;           1 = dual line block sequence
numbering APPLIES

; on exit
;   a = destroyed
;   b = destroyed
;   x0 = destroyed
;   x1 = destroyed
;   y0 = destroyed
;   y1 = destroyed

section blkseqnums
xdef seqnums
xdef nxtseq,seqnum
xdef frmendpos ;bit position of the true end of the frame
xdef audendpos ;bit position of end of MUSICAM frame
xdef bsnendpos ;bit position for block sequence number
xdef spltrte
xdef spltbnd
xdef spltmxsubs
xdef spltpaddiff

;***** (start)  SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****

        xdef nxtsc
        xdef nxtcurwd
        xdef nxtstrt

;***** (end)    SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****

        org yhe:
stbitsallo_yhe

;define the low order 12 bits of the coded block sequence numbers
for
;   dual output line bit allocation:
;   bits 0-9 contain the echoed block sequence number in the
;   range of 00 thru 31 (each bit is duplicated
;   for the even/odd line transmission)

```

bit 10 = 1 for output over one of the 2 lines  
 bit 11 = 0 for output over the other of the 2 lines

seqnums			
dc	S000400	;01 - 00 00 00 00 00	sequence no. 0 =
00000)			
dc	S000403	;01 - 00 00 00 00 11	sequence no. 1 =
00001)			
dc	S00040c	;01 - 00 00 00 11 00	sequence no. 2 =
00010)			
dc	S00040f	;01 - 00 00 00 11 11	sequence no. 3 =
00011)			
dc	S000430	;01 - 00 00 11 00 00	(sequence no. 4 =
00100)			
dc	S000433	;01 - 00 00 11 00 11	(sequence no. 5 =
00101)			
dc	S00043c	;01 - 00 00 11 11 00	(sequence no. 6 =
00110)			
dc	S00043f	;01 - 00 00 11 11 11	(sequence no. 7 =
00111)			
dc	S0004c0	;01 - 00 11 00 00 00	(sequence no. 8 =
01000)			
dc	S0004c3	;01 - 00 11 00 00 11	(sequence no. 9 =
01001)			
dc	S0004cc	;01 - 00 11 00 11 00	(sequence no. 10 =
01010)			
dc	S0004cf	;01 - 00 11 00 11 11	(sequence no. 11 =
01011)			
dc	S0004f0	;01 - 00 11 11 00 00	(sequence no. 12 =
01100)			
dc	S0004f3	;01 - 00 11 11 00 11	(sequence no. 13 =
01101)			
dc	S0004fc	;01 - 00 11 11 11 00	(sequence no. 14 =
01110)			
dc	S0004ff	;01 - 00 11 11 11 11	(sequence no. 15 =
01111)			
dc	S000700	;01 - 11 00 00 00 00	(sequence no. 16 =
10000)			
dc	S000703	;01 - 11 00 00 00 11	(sequence no. 17 =
10001)			
dc	S00070c	;01 - 11 00 00 11 00	(sequence no. 18 =
10010)			
dc	S00070f	;01 - 11 00 00 11 11	(sequence no. 19 =
10011)			
dc	S000730	;01 - 11 00 11 00 00	(sequence no. 20 =
10100)			
dc	S000733	;01 - 11 00 11 00 11	sequence no. 21 =
10101)			
dc	S00073c	;01 - 11 00 11 11 00	sequence no. 22 =
10110)			
dc	S00073f	;01 - 11 00 11 11 11	sequence no. 23 =
10111)			
dc	S0007c0	;01 - 11 11 00 00 00	sequence no. 24 =
11000)			



```

        dc  S0007c3          ;01 - 11 11 00 00 11 sequence no. 25 =
110011
        dc  S0007cc          ;01 - 11 11 00 11 00 sequence no. 26 =
110101
        dc  S0007cf          ;01 - 11 11 00 11 11 sequence no. 27 =
110111
        dc  S0007f0          ;01 - 11 11 11 00 00 sequence no. 28 =
111001
        dc  S0007f3          ;01 - 11 11 11 00 11 sequence no. 29 =
111011
        dc  S0007fc          ;01 - 11 11 11 11 00 sequence no. 30 =
111101
        dc  S0007ff          ;01 - 11 11 11 11 11 sequence no. 31 =
111111
endsequence
nxtseq      ds  1          ;address of next
seqnum      ds  1          ;block sequence number to set A-bit
frmendpos ds  1          ;bit position of the true end of the frame
audendpos ds  1          ;bit position of end of MUSICAM frame
bsnendpos ds  1          ;bit position for block sequence number
spltrte ds  1          ;split mono frame bit rate code for frame hdr
spltbnd ds  1          ;split mono frame bit rate code for bandwidth
spltmxsubs ds  1          ;split mono frame MAXSUBBANDS
spltpaddiff ds  1          ;frame padding calc: DIFF @ sample/bit
rates

;***** (start) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****

nxtsc      ds  1          ;y:<sc value to start next frame
nxtcurwd ds  1          ;y:<curwd partly formatted word-start next
frame
nxtstrt ds  1          ;y:<frmstrt buffer address to start next
frame

;***** (end) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****

endbitsallo_yhe
endsec

bitsfree

;pad 0 bits thru the end of the coded frame including any client
trailing bits

        move y:frmendpos,r0          ;bit count thru CLIENT bits
;set flag for reed solomon, if reed solomon, skip the frame flush:
        move #reedsolomon,r4          ;addr of the flag
        jsr flushframe          ;pad frame with zeroes

```

```

        tst a                ;check for an overshoot????
        jge _free_00        ;OK, see if we have a split frame

;OVERSHOOT ERROR!!! this case should not occur

        ON_BITALLOC_LED_CD    ;!!! error we've overshoot

;!!!debug: dump the frame in question (pull of the '/' from next
;line!

        jsr dumpdata

        jmp _free_90        ;done with bad frame

_free_00

;see if split frame applies, if not, we should have coded all bits
in frame

        jclr #SPLIT_MODE,y:<stereo,_free_90    ;if not split, chk end
of frame

;if NOT a split mono frame, output the block sequence number

        jclr #SPLIT_MONO_FRAME,y:<stereo,_free_20

;***** (start) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****
;***** (start) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****

;format the block sequence number for the last word of the frame
buffer

        move y:nxtseq,r1    ;block sequence number to output
        move #31,m1        ;circular buffer thru blk seq num tbl
        nop
        move y:(r1)+,x1    ;get blk seq num, incr for next frame
;!!!dbg move y:(r1),x1 ;!!!dbg keep the same BSN at end of frame
        move x1,y:seqnum
        move r1,y:nxtseq    ;save for next frame blk seq num

;test if one of the receiving lines is down and this frame is a
split frame

        TST_CLR_TRAN_A_ERROR_CD,_split_00    ;NOT A-bit set to 1
        bset #A_BIT_OFFSET_ODD,y:seqnum    ;set bit for line 1
        bset #A_BIT_OFFSET_EVEN,y:seqnum    ;set bit for line 2

_split_00

; determine word and bit offsets for the end of the entire frame

        move y:<outsize,m0    ;set for circular buffer control

```

```

        move y:<frmsc,x0          ;set frame start address bit offset
        move #>24,a              ;set number bits in a word
        move #>24,y1             ;set number bits in a word
        sub x0,a y:<bitsfrm,b     ;set bit count for frame in 1st word
                                   ; & get bit count for current frame
        cmp y1,a y:<frmstrt,r0    ;see if entire 1st word of frame
                                   ; & set frame start address
        jeq _splt_10             ;if word fits, go right into loop

;only part of 1st word contains the frame,
; a. sub bits from entire frame bit count
; b. increment address counter
        sub a,b (r0)-

_splt_10
;adjust address to end of the frame as per 24 bits per word giving
the
;word address and bit count to start the next frame

        cmp y1,b                ;see if reached last word
        jlt _splt_20            ;if so, set eoframe word & bit offsets
        sub y1,b (r0)+
        jmp _splt_10

_splt_20
        move b,y:nxtsc          ;bit offset start next formatted frame
        tst b r0,y:nxtstrt      ;if bit offset not zero, next addr
sec
                                   ; & set buffer addr start next frame
        jne _splt_25            ;if offset 0, incr addr start next frame
        move (r0)-              ;back up addr to end of current frame

_splt_25
;set up the end word of the current frame with the
; left justified block sequence number
;the end bits in the frame will shift the end word back right

        bclr #4,y:<not_appl      ;indicate end word of frame NOT done
        move y:seqnum,b         ;get the block sequence number

        move #24-BLOCK_SEQ_NUM_BITS,r2 ;set num bsn bits to roll left
;left justify the block sequence number

        do r2,_splt_30
        rol b                   ;roll left up to 1st data bit

_splt_30
;position at the end of the formatted frame and the end of the
frame buffer

```

```

; prior to the block sequence number

    move y:<outsize,m1      ;set circular buffer ctl for source
    move y:<sc,a            ;numb partial formatted source bits
    move #0,r3             ;no bits to rotate from source yet
    tst a    r6,r1         ;see if any bits partially formatted
                                ; & set the source start address
    jeq _splt_40           ;no partial bits, start at last insert

;the end of the frame is partially formatted in y:curwd for y:sc
bits

    move y:<sc,r3           ;set bit counter in partial format word
    move y:<curwd,a         ;get right justified part formatted
word

_splt_40

;see if source is ready to get the previous word

    move a1,b0             ;save current shifted word
    move r3,a              ;get the bit counter
    tst a    b0,a1         ;test for zero & restore shifted word
    jne _splt_50           ;is still bits to go, continue

;test if we just finished the 1st word in the source and if so,
; we're done, output the 1st word of the frame and continue

    move y:<frmstrt,x1      ;backed to the start of the frame?
    move r1,a              ;last word addr eq to frame start addr
    cmp x1,a (r1)-         ;test equal, & back up to previous
word
    jeq _splt_120          ;if eq, we're done

    move #24,r3            ;start with a new word

;see if this new word to be processed is the frame start word.
; if so, adjust for the bit offset to the start of the frame SYNC

    move r1,a              ;see if new word addr eq to frame start
    cmp x1,a #>24,a        ;test if at the 1st word of frame
                                ; & set for bit count if it is 1st word
    jne _splt_45           ;if not 1st word, get the new word
    move y:<frmstc,x1       ;get frame start address bit offset
    sub x1,a              ;calculate bits in the 1st frame word
    move a,r3              ;and move to the source word bit ctl

_splt_45

;take the next word to be processed

    move y:(r1),a          ;get previous word

_splt_50

```

294

BAD ORIGINAL

```

        move r3,-          ;decrement shifted bit
        ror a              ;move source bit to carry bit
        jcs _splt_61       ;see if carry bit is a 1
        bclr #10,y:<not_appl ;flag that carry bit is 1
        jmp _splt_70

_splt_61
        bset #10,y:<not_appl ;flag that carry bit is 1

_splt_70

;output the carry bit twice for line 1 and line 2
        do #2,_splt_110

;see if destination is ready to get the previous word

        move b1,a0          ;save current shifted word
        move r2,b           ;get the bit counter
        tst b a0,b1         ;test for zero & restore shifted word
        jne _splt_30        ;is still bits to go, continue

;see if this is the end word of the frame and if so,
; set the y:nxtcurwd for the start of the next frame
; and make any adjustments for the current frame

        jset #4,y:<not_appl,_splt_78
        bset #4,y:<not_appl ;indicate end word handled
        move b1,y:nxtcurwd ;start of the next frame formatted wd
        move y:nxtsc,b      ;get the start bit for the next frame
        tst b b,r2          ;see if zero
        ; & set the value to roll left
        jeq _splt_76        ;if zero, end word is all set

;get current buffer end word roll left to abut the previous frame
start bit
; with the end bit of current frame

        clr b               ;zero the b register
        move y:(r0),b0      ;get end word from frame buffer
        do r2,_splt_72      ;shift the end word bits in b0
        asl b               ; so they are left justified

_splt_72

;roll the end word to isolate the end bits for the frame buffer

        move a0,b1          ;restore formatted end word
        do r2,_splt_74      ;shift the nxtcurwd bits into b0
        asr b               ; so they are left justified

_splt_74

;store the reformatted end word (and start of previous frame)

```

```

; and restore the formatted end word with r2 set according to
; shift

    move b0,y:(r0)-          ;store formatted end word back in buf
                                ; & decrement address for next word o/p
    move a0,b1                ;restore formatted end word
    jmp _split_80             ;continue by inserting bits

_split_76
;no bits needed to be shifted, the end word is all set

    move a0,b1                ;restore formatted end word

_split_78
;store reformatted word in the frame buffer

    move #24,r2                ;start with a new word
    move b1,y:(r0)-           ;put new word out to buffer & back up

_split_80
;either clear or set the carry bit

    jset #10,y:(not_appl,_split_90) ;is carry bit is to be restored
to 1
    andi #SFE,CCR             ;set the carry bit to 0
    jmp _split_100

_split_90
    ori #S01,CCR              ;set the carry bit to 1

_split_100
;count the bits inserted and insert the bit

    move (r2)-                 ;decrement shifted bit
    ror b                     ;move carry bit into word

_split_110
;go back for the next bit from the source

    jmp _split_40

_split_112
;see if partially formatted 1st word in the frame
; if so, right adjust the partial word and and it with end of
previous frame

    move #>24,a                ;set bits per word
    move r2,x0                 ;get shifted bit count downer

```

```

sub x0,a          ;see if any bits shifted in to b1
cmp y1,a a,r2     ;test for no bits partially formatted
                  ; & set the shift bit counter
;eq _splt_140      ;if no bits to go, continue

move y:(r0),a     ;get the word at frame start

;right align the last word in previous frame

do r2,_splt_130
asr a             ;shift right up to 1st data bit

_splt_130

;now about the frame start bits (a0) with the end bits of previous
frame (a1)

move b1,a0        ;partial formatted word to a0

;shift left to align the last word in previous frame
; with start bits of current frame in a1

do r2,_splt_135
asl a             ;shift left up to 1st data bit

_splt_135

;now put the reformatted word in the proper register

move a1,b         ;a1 = end of prev start of current frame

_splt_140
move b1,y:(r0)    ;output new 1st word out to buffer
move #-1,m0       ;reset to linear buffer control
move #-1,m1       ;reset to linear buffer control
jmp _free_90      ;set addr for skf checksums - next frame

;***** (end) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****
;***** (end) SPLIT_MONO_FRAME ONLY (to make 2nd copy of frame):
;*****

;SPLIT_MODE frame but NOT a SPLIT_MONO_FRAME position for the BSN

_free_20
move y:nxtseq,r1   ;block sequence number to output
move #31,m1        ;circular buffer thru blk seq num tbl
move #>BLOCK_SEQ_NUM_BITS,x0 ;number of bits for block seq
num
move y:(r1)-,x1    ;get blk seq num, incr for next frame
;!!!dbg move y:(r1),x1 ;!!!dbg keep the same BSN at end of frame
move x1,y:seqnum   ;store the selected sequence number
move r1,y:nxtseq   ;save for next frame blk seq num
move #-1,m1        ;restore to linear buffer ctrl

```



```
;test if one of the receiving lines is down and this frame is a
split frame
```

```
TST CLR_TRAN A ERROR CD, _free_30 ;NOT A-bit set to 1
bset #A_BIT_OFFSET_ODD,y:seqnum ;set bit for line 1
bset #A_BIT_OFFSET_EVEN,y:seqnum ;set bit for line 2
```

```
_free_30
move #BLOCK_SEQ_NUM_BITS,n4 ;number of bits for block seq
num
move y:seqnum,y0 ;block seq number to output
jsr setvalue ;add blk seq num in last word
```

```
_free_90
;set address for the next frame's scale factor checksums
; a. get bit count for CLIENT bits and the scale factor checksums
; b. get bit count to the end of the formatted frame (block seq
number)
; c. if this is a combined mode split mono frame, double the bit
count
; for CLIENT and checksums
; d. determine word address and bit count to come back and insert
the
scale factor checksums in the already coded frame
```

```
move #>CLIENT_TRAILING_BITS,r0
move #>NSKFCRCBITS*NUMSKFCKSUMS,n0
move y:bsnendpos,b ;bit count to end of frame
move (r0)+n0 ;bits for client + checksums
move r0,n0 ;if need to be doubled
```

```
;set flag for reed solomon (if reed solomon, scale factor crc next
to insert)
```

```
move #reedsolomon,r4 ;addr of the flag
```

```
;test for a split mono frame in order to double the bit count
```

```
jclr #SPLIT_MONO_FRAME,y:<stereo,_free_92 ;if not, continue
move (r0)+n0 ;double bit count
```

```
_free_92
```

```
;for reed solomon, save current word address and bit offset for
the insertion of the next frame's scale factor checksums
```

```
jclr #0,y:(r4),_free_93 ;if not reed solomon, continue
move r6,x:skfrcwd ;word addr after and data & client
bits
move y:<sc,x0 ;get bit offset into next word to x/p
move x0,x:skfrcbt ;bit offset after and data & client
bits
```



```

;now flush the rest of the frame with zero bits
        move #not_appl,r4          ;use this addr for the flush to be
done    bclr #0,y:<not_appl        ;make sure bit is zero for flush
        move y:reedendpos,r0      ;bit count thru rest of the
buffer
;pad the remainder of the frame with zero bits
        jsr flushframe            ;pad frame with zeroes
        tst a                     ;check for an overshoot????
        jge _free_97             ;OK, skip info for next frame skf crc's
;OVERSHOOT ERROR!!! this case should not occur
        ON_BITALLOC_LED_CD        ;!!! error we've overshoot
;!!!debug: dump the frame in question (pull of the '/' from next
line)
        jsr dumpdata
        jmp _free_97             ;skip info for next frame skf crc's
_free_93
;subtract the bits for client and checksums from the end of frame
bit count
        move r0,x0                ;bit count client and checksums
        sub x0,b y:<frmstrt,r0    ;sub from end frame bit count
                                   ; & curr frame start address
;set start of frame address and circular buffer ctl in order to
;calculate address and bit offset to store the next frame's
checksums
        move y:<frmstrt,r0        ;curr frame start address
        move y:<outsize,m0        ;circ buffer control
;get bits partially formatted in the 1st word
;account for the 1st partially formatted word of the frame
        move #>24,a               ;to determine bits in 1st word
        move y:<frmsc,x0          ;get bit offset start frame
        sub x0,a #>24,x0          ;calc bits in 1st word of frame
        sub a,b (r0)+             ;sub 1st wd patrial bits
                                   ; & increment the address
;loop subtracting 24 bits per word from end of frame bit count and
incrementing
; the address to reach the place for the next frame's scale factor
checksums

```



```

_free_94
    cmp     x0,b           ;24 bits vs current bit counter
    jlt     _free_96       ;if less, we reached the address

;subtract 24 bits from end of frame bit counter
    sub     x0,b,r0)-      ;sub 24 bits from curr bit count
                        ; & increment the address
    jmp     _free_94       ;continue looping

_free_96
;save the calculated address and the bit offset to code the next
frame's crc's
    move    r0,x:skfcrwd    ;save address
    move    b,x:skfcrbt     ;save bit offset
    move    #-1,m0         ;restore linear buffer ctl

_free_97
;clear the flag that this frame is a split mono frame
    bclr    #1,x:private

;if this is not split mono frame, go to validate the proper end of
frame
    jclr    #SPLIT_MONO_FRAME,y:<stereo,_free_98

;set the flag that this frame is a split mono frame
    bset    #1,x:private

;doing a split mono frame: set controls for starting the next frame
    move    y:nxtsc,x0      ;set the y:<sc bit offset to start
    move    x0,y:<sc        ;store bit offset to start y:<curwd
    move    y:nxtcurwd,x0   ;get the y:<curwd formatted word
    move    x0,y:<curwd     ;store 1st partial formatted word
    move    y:nxtstrt,x0    ;get frame buffer start address
;!!!!dbg    move    x0,y:<frmstrt    ;store frame start address
    move    x0,y:<frmnext    ;store frame start address
    jmp     _free_100       ;we're done

_free_98
;ensure that we have coded to the end of the frame
    move    y:<bitsfrm,x0    ;get true frame length in bits
    move    y:<bitscnt,a     ;get count of bits output so far
;!!!!dbg    cmp     x0,a,r6,y:<frmstrt ;these should be equal
    cmp     x0,a,r6,y:<frmnext ;these should be equal
                        ; & save for start of next frame

```

300

BAD ORIGINAL



```
beq _free_100      ;OK!! all went according to plan
;FRAME ENCODE ERROR!!! this case should not occur
CN_BITALLOC_LED_CL      ;!!! error we've overshot
;!!!debug: dump the frame in question (pull of the ';' from next
line:
;    jsr  dumpdata
;
;    move #framebuf,r0      ;start pointers over
;    move y:<outmus.n0      ;to advance 1 frame
;!!!dbg  move r0,y:<frmstrt ;at beginning of the buffer
;    move r0,y:<frmnext     ;at beginning of the buffer
;    move (r0)+n0           ;address of the second frame
;    move r0,y:<oprptr      ;output read pointer 1 frame ahead
_free_100
    rts
```



© 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.

UXCCCB\fftr16b.asm

Radix 2, In-Place, Decimation-In-Time FFT fast

Last Update 13-Sept-91 Version 1.0

```
fftr16b macro points,data,coef,coef1,dacol
fftr16b ident 1.0
```

Radix 2 Decimation in Time In-Place Fast Fourier Transform Routine

Real input and complex output data  
 Real data in X memory  
 Imaginary data in Y memory  
 Normally ordered input data  
 Bit-reversed complex output data  
 Coefficient lookup table  
 -Cosine values in X memory  
 -Sine values in Y memory  
 -fast index search in X & Y memory

Macro Call - fftr16b points,data,coef,coef1,dacol

points number of points (16-32768, power of 2)  
 data start of data buffer  
 coef start of sine/cosine table  
 dacol start of index table

Alters Data ALU Registers

x1	x0	y1	y0
a2	a1	a0	a
b2	b1	b0	b

Alters Address Registers

r0	n0	m0
r1	n1	m1
r2	n2	m2
r3	n3	
r4	n4	m4
r5	n5	m5
r6	n6	m6
	n7	

Alters Program Control Registers

pc	sr
----	----

Uses 6 locations on System Stack

Latest Revision - 13-Sept-90

```
move #data,r1      ;initialize input pointer
move #points/4,n0   ;initialize input and output pointers offset
move n0,n1          ;initialize input pointers offset
move n0,n4          ;initialize output pointers offset
move #1,n2          ;initialize groups per pass
move #coef,r6       ;initialize sine/cosine input pointers
move #1024,n6       ;relative address
```

302

BAD ORIGINAL

```

move #cvi @log(points) @log(2)-3, r3
; Do first and second Radix 2 FFT passes, combined as 4-point butterflies

lua (r0)-n0, r1          ; initialize input B pointer
move #points-1, m0        ; initialize address modifiers
move m0, m1              ; for modulo addressing
move m0, m4
lua (r1)-n1, r4          ; initialize output C pointer
move m0, m5
move x: (r0), x0
lua (r4)-n4, r5          ; initialize output D pointer
do n0, _twopass
  tfr x0, a              x: (r1), y1
  tfr y1, b              x: (r4), y0
  add y0, a              x: (r5), -x1
  add x1, b              ; ar+cr
  add a, b              ; br+dr
  sub1 b, a              ; ar'=(ar+cr)-br-dr
  tfr x0, a              ; br'=(ar+cr)-br-dr
  sub y0, a              a, x: (r1)-
  sub y1, b              x1, b
  move x: (r0), x0        ; cr'=ar-cr
                        ; ci'=dr-br
  move x: (r0), x0        b, y: (r4)-
_twopass
  move (r0)+n0

; Do the complex FFT using butterfly kernel to 2nd last pass

do #cvi(@log(points)/@log(2)-3), _t_end
  move r0, n3              ; save the beginning address
  move #dacol, r2          ; reset the index table
  move n0, b1
  lsr b
  move b1, n0
  move n0, n7              ; save the input offset
  do r3, _toendpass2
    move r0, r4            ; initialize output pointers
    lua (r0)+n0, r1        ; initialize input B pointer
    move n0, n1            ; initialize all the input output offset
    move n0, n4
    move n0, n5
    lua (r1)-, r5          ; initialize output D pointer
    do n2, _endgroup        ; calculate the group FFT
      move y: (r2)+, r6
      move x: (r5), a        y: (r0), b
      move (r6)-n6
      move x: (r1), x1        y: (r6), y0
      move x: (r6), x0
      do n0, _bufknl        ; Kernel FFT processing
        mac x1, y0, b        y: (r1)-, y1
        macr -x0, y1, b      a, x: (r5)-, y: (r0), a
        subl b, a            x: (r0), b        b, y: (r4)
        mac -x1, x0, b        x: (r0)-, a        a, y: (r5)
        macr -y1, y0, b      x: (r1), x1
        subl b, a            b, x: (r4)-, y: (r0), b
      _bufknl
      move a, x: (r5)-n5      y: (r1)-n1, y1
      move x: (r0)-n0, x1      y: (r4)-n4, y1
    _endgroup
  move n3, r0              ; reset the beginning address for next pass

```

```

        move n1,n1          ;update the new group number
        lsr b n2,a1
        lsl a n1,n2
        move a1,n2
    _endpass2
; Do last pass for all the complex FFTs
    lua r0)-,r1             ;initialize input B pointer
    move #2,n0              ;initialize FFT elements in each group
    move r0,r4              ;initialize output C pointer
    move n0,n1              ;initialize all input/output offset
    move n0,n4
    move n0,n5
    move r1,r5              ;initialize output D pointer
    move y:(r2)-,r6
    move y:(r3),b
    move (r5)-n6
    do n2,_endgroup1        ;each group is just one kernel process
        move x:(r1),x1      y:(r5),y0
        move x:(r6),x0
        mac x1,y0,b          y:(r1)-n1,y1
        macr -x0,y1,b        y:(r0),a
        subl b,a             x:(r0),b          b,y:(r4)
        mac -x1,x0,b          x:(r0)-n0,a        a,y:(r5)
        macr -y1,y0,b        y:(r2)-,r6
        subl b,a             b,x:(r4)-n4          y:(r0),b
        move a,x:(r5)-n5
        move (r6)-n6
    _endgroup1
; Do the half upper real part's FFT
    move #data,r0
    move n7,n0
    move n0,n1
    move n0,n4
    lua (r0)+n0,r1
    move #1,n2
    lua (r1)-n1,r4
    move (r3)-
    lua (r4)-n4,r5
    move x:(r0),a
    move x:(r1),y0
    do n0,_uponep
        add y0,a             a,b
        subl a,b             a,x:(r0)-          ;ar'=ar+cr
        move b,x:(r1)+       ;br'=ar-cr
        move x:(r5)-,b1
        neg b                 x:(r1),a
        move b,y:(r4)-       ;ci'=-dr
        move x:(r1),y0
    _uponep
    move r1,-n1
    t_end
; Do the beginning four point FFT at last pass
    move #data,r0
    move #2,n0

```

304

BAD ORIGINAL

```

move =daddl,r3
move x1,r1,-,x0
lfr x0,a          x1:r0,y0
sub y0,a          y1:r1,r6      ;br'=ar-cr
move a,x1:r0,-n0
move x1:r1,-,x1
neg b              r6=-n6      ;dr'=-dr
move a1,y1:r0-n1

```

Complex 2 point FFT for last pass

```

lra r0,-r1
move y:r0,b
move x1:r1,x1      y:r6,y0
move x1:r6,x0      y:r1,y1
mac x1,y0,b         y:r0,a
macr -x0,y1,b       b,y:r0
subl b,a            ;c1=a1+bicos-brsin
move x1:r0,b        ;d1=2a1-c1
mac -x1,x0,b        a,y:r1
macr -y1,y0,b       x:r0,a
subl b,a            ;cr=ar+brcos+bsin
move a,x1:r1        ;dr=2ar-cr
endm

```



```
opt    fo.cax,mex
```

```
© 1994, Copyright Corporate Computer Systems, Inc. All rights reserved.
```

```
UXCODE\setvalue.asm
```

```
This routine is used to output bits to the output bit buffer.  
The basic idea is to look at 3 different cases:
```

1. The new bits fit entirely in the current word with room to spare.
2. The new bits fit exactly in the current word.
3. The new bits exceed the available room in the current word and thus the current word is filled and a new word is started.

```
title    'Set Value'
```

```
on entry
```

```
    r6 = address of the next word to the output buffer in memory  
    y0 = value to output (right justified)  
    n4 = number of bits to output (1-16)
```

```
    y:<curwd = current word being formed for the frame  
    y:<sc = current bit position in current word being formed for the frame  
    y:<bitscnt = count of bits put to the frame
```

```
on exit
```

```
    a = destroyed  
    b = destroyed  
    y0 = destroyed  
    y1 = destroyed  
    r4 = destroyed
```

```
    r6 = updated for next word in output buffer (OutData)
```

```
    y:<curwd = updated with bit changes last inserted  
    y:<sc = updated bit position into the current word being formatted  
    y:<bitscnt = update count of bits put to the frame
```

```
include '..\uxcode\setvalue.mac'
```

```
section ytables  
xdef    shifttbl  
xdef    ldshfttbl
```

```
org      yhe:  
stsetvalue_yhe
```

```
shifttbl
```

dc	5000000	;place holder
dc	5400000	;shift value for 1 bit
dc	5200000	;shift value for 2 bit
dc	5100000	;shift value for 3 bit
dc	5080000	;shift value for 4 bit
dc	5040000	;shift value for 5 bit
dc	5020000	;shift value for 6 bit
dc	5010000	;shift value for 7 bit
dc	5008000	;shift value for 8 bit
dc	5004000	;shift value for 9 bit
dc	5002000	;shift value for 10 bit
dc	5001000	;shift value for 11 bit

306

BAD, ORIGINAL





```

dc      5000300      ;shift value for 12 bit
dc      5000400      ;shift value for 13 bit
dc      5000200      ;shift value for 14 bit
dc      5000100      ;shift value for 15 bit
dc      5000080      ;shift value for 16 bit

ldshftbl
dc      5000000      ;place holder
dc      5000001      ;shift left 1 bit
dc      5000002      ;shift left 2 bits
dc      5000004      ;shift left 3 bits
dc      5000008      ;shift left 4 bits
dc      5000010      ;shift left 5 bits
dc      5000020      ;shift left 6 bits
dc      5000040      ;shift left 7 bits
dc      5000080      ;shift left 8 bits
dc      5000100      ;shift left 9 bits
dc      5000200      ;shift left 10 bits
dc      5000400      ;shift left 11 bits
dc      5000800      ;shift left 12 bits
dc      5001000      ;shift left 13 bits
dc      5002000      ;shift left 14 bits
dc      5004000      ;shift left 15 bits
dc      5008000      ;shift left 16 bits

endsetvalue_yhe
endsec

section highmisc
xdef    svb1
xdef    svn4

org      xhe:
stsetvalue_xhe

svb1     ds      1
svn4     ds      1

endsetvalue_xhe
endsec

org      pli:

setvalue

;set up for the setvalue macro

SETUP4SETVALUE
; move    y:<sc,y1      ;get # of bits left in current word
; move    r4,b          ;set # of bits
; clr     a              ;prepare a register
;          y:<bitscnt,r4 ; & get # of bits used so far
; move    y0,a0          ;put values into proper register

;use the setvalue macro

SETVALUE

rts

```

307

BAD ORIGINAL

no: list

© 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.

UKCODE setvalue.mac

This routine is used to output bits to the output bit buffer.

The basic idea is to look at 3 different cases:

1. The new bits fit entirely in the current word with room to spare.
2. The new bits fit exactly in the current word.
3. The new bits exceed the available room in the current word and thus the current word is filled and a new word is started.

y:curwd = current word being formed for the frame

y:sc = current bit position in current word being formed for the frame

y:bitscnt = count of bits put to the frame

on entry

b = number of bits to output (1-16, same as n4)

r6 = address of the next word to the output buffer (y memory)

a2 = 0

a1 = 0

a0 = value to output (right justified)

y1 = y:sc (# of bits left in current word)

r4 = y:bitscnt (number of bits output up to this call)

n4 = number of bits to output (1-16)

on exit

a = destroyed

b = destroyed

y0 = destroyed

y1 = destroyed

r4 = destroyed

n4 = MUST BE SAFE ACROSS THIS CALL

x0 = MUST BE SAFE ACROSS THIS CALL

x1 = MUST BE SAFE ACROSS THIS CALL

r6 = updated for next word in output buffer (OutData)

y:curwd = updated with bit changes last inserted

y:sc = updated bit position into the current word being formatted

y:bitscnt = update count of bits put to the frame

SETUP4SETVALUE macro

.....  
: The next 4 lines should be in quantize.asm, setvalue.asm, ...  
: NOTE: quantize.mac already leaves the value in a0  
: They should be removed from this routine.

.....  
move y:sc,y1 ;get # of bits left in current word  
move n4,b ;set # of bits  
clr a y:bitscnt,r4 ;get # of bits used so far  
move y0,a0 ;put value into proper register  
.....  
endm

SETVALUE macro

add y1,b =>24,y1

;add bits to bsp to offset

308

BAD ORIGINAL

```

        move      r4,-n4                ; & set compare to 24 bits/word
                                         ; update total bits used so far

; see if this value will fit totally in current output word

        cmp      y1,b      r4,y:<bitscnt ; see if new value fits
                                         ; & save new total bit count

        move     #ldshfttbl,r4          ; get shift table address

        flt      _setv_70              ; fits within current word
        leq      _setv_60              ; exactly fits

; the current value is too big so we must do it in 2 parts.
; part 1 - do the part which fits in the remaining bits.
; part 2 - do the part which is left over.

; NOTE: b2 and b0 will be zero as a
; result of this operation

; find the number of bits left in the current word

;!!!!N/A move     a0,y0                ; save bits to output in a save register
        move     y:<sc,a              ; get # of bits used in current word
        sub      y1,a      x0,x:svb1  ; get # of bits which just fit
                                         ; save x0 register
        neg      a      y:<curwd,x0    ; make -
                                         ; get current word we are working on

        move     n4,x:svn4            ; save the # of bits

        move     a,n4                ; save # for this pass
        move     y0,y:<curwd          ; save as the new current word. Note that
                                         ; we don't need to mask off the unused
                                         ; upper bits since the word will be
                                         ; shifted left soon.

; Move the current word left to make room for the new bits.
; The current word will be completely full after completing this section.

        move     y:(r4+n4),y1         ; get shift value
        mpy      y1,x0,a #>24,y1      ; shift old bits for new value.

; Now move the msb's of the input right to fit into the lsb of the
; current word.

        sub      y1,b      x:svb1,x0  ; compute # of bits in next word
                                         ; & restore x0
        move     b,n4                ; number of bits leftover
        move     #shiffttbl,r4        ; address of right shift table
        move     a0,a                ; move to correct register
        move     y: r4-n4,y1          ; get shift value
        mac      y0,y1,a b,y:<sc      ; shift input word right into a1
                                         ; & insert new value at end of new curwd
        move     a1,y:(r6)+           ; output word to the buffer

        move     x:svn4,n4            ; restore the # of bits to output

        jmp      _setv_90             ; and we are done

```



; The current value just fits

```
_setv_60
    move    y:<curwd,y0          ;get current word in output buf
    move    y:(r4+n4),y1        ;get left shift value
    mac     y1,y0,a b0,y:<sc      ;shift old bits for new value
                                   ; & set bits used in current word to 0
    move    a0,y:(r6)-          ;output word to the buffer
    jmp     _setv_90
```

; this is the case when the value fits in the current word

```
_setv_70
    move    y:<curwd,y0          ;get current word in output buf
    move    y:(r4+n4),y1        ;get left shift value
    mac     y1,y0,a b,y:<sc      ;shift old bits for new value
                                   ; & update bits used in current word
    move    a0,y:<curwd          ;save current output word
```

\_setv\_90

endm  
list

310

BAD.ORIGINAL



```

opt    fo

; c 1991. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; UXC00DE\setcrc.asm

title  'Set the checksum word'

; These routines maintain the checksum protection portion in the frame
; setcrc: initializes the checksum portion in the frame by inserting 16
;         0 bits and thereby saving space for the calculated result
;         the 16-bit check sum after the header and before the bit allocations
;         info in bits 32-47 of the frame.
; setcrc: calls the routine to calculate the check sum and outputs
;         the 16-bit check sum after the header and before the bit allocations
;         info in bits 32-47 of the frame.

; on entry
;     r6 = current offset in output array
;     y:sc = shift count

; on exit
;     a = destroyed
;     b = destroyed
;     y0 = destroyed
;     y1 = destroyed
;     r4 = destroyed
;     n4 = destroyed

include 'def.asm'

section lowmisc
xdef    frmaddr
xdef    frmisc
xdef    crcaddr
xdef    crcsc

org     yli:
stsetcrc_yli

;;frmaddr    ds      1                ;address of start of channel frame heade
frmisc    ds      1                ;bit offset into word to start channel frame
crcaddr    ds      1                ;address of start of frame's CRC checksum
crcsc      ds      1                ;bit offset of start of frame's CRC checksum

endsetcrc_yli
endsec

section highmisc
xdef    crcbits
xdef    crcold
xdef    chksum

org     xhe:
stsetcrc_xhe

crcbits    ds      1                ;NEW: accum span of bits for CRC-16 rtn
crcold     ds      1                ;OLD: fixed span of bits for CRC-16 rtn
chksum     ds      1                ;save calculated checksum

```

311

BAD, ORIGINAL

```
endsetcrc_xhe
endsec
```

```
org phe:
```

```
clearc
```

```
;this subroutine clears the checksum in the frame buffer
; and saves its address in the frame buffer
```

```
move    r6,y:<crdaddr      ;save address for inserting crc checksum
move    y:<sc,x1            ;current bit offset for CRC checksum
clr     a      x1,y:<crsc   ;zeros for the checksum
                        ; & save the CRC starting bit offset
move    a,y0              ;value to be output
move    #NCRCBITS,n4      ;number of bits
move    #CRC_BITS_A,r1    ;insert bit cnt for header & checksum
move    r1,x:crclbits     ;init bit ctr for span covered by CRC-16
jsr     setvalue          ;output the value

rts
```

```
setcrc
```

```
; x:crclbits = accumulator of bits covered by CRC-16 routine
```

```
;this subroutine calls the calculate checksum routine
```

```
; and then inserts the result into frame buffer
```

```
; a. set starting address and bit offset of this channel frame header
```

```
; b. calculate the offset to start the checksum calculation
```

```
;; move    y:<frmaddr,r0      ;get address of start of frame buffer
move    y:<frmstrt,r0        ;get address of start of frame buffer
move    m6,m0               ;set circular buffer control
move    y:<frmsc,a           ;get the starting bit offset of frame

move    =>CRC_SUM_BIT_OFFSET,x1 ;calculate msb position from which to
                        ; start calculating the checksum
add     x1,a      #>24,x1    ;set offset to start checksum calculate
                        ; & to check overflow to next word
cmp     x1,a               ;see if offset to start in next word
jlt     _scrc_a            ;if less, we're all set
```

```
;adjust address up 1 position and adjust bit offset to start for CRC-16 run
```

```
sub     x1,a               ;bits for 1 word to adjust bit offset
move    (r0)+              ;increment start word address
```

```
_scrc_a
```

```
move    a,x1               ;bit offset to start checksum calculate
move    =>CRC_VALUE,y1      ;set the checksum divisor
```

```
;for ISO old or new CRC-16 controls:
```

```
; set the checksum seed value and the number of bits covered by the checksum
```

```
jset    #CRC_OLD_vs_NEW,y:<stereo,_scrc_00
```

```
move    x:crcold,r1        ;get OLD bit ctr for span over CRC-16
move    #0,x0              ;OLD: seed the checksum with 0's
```

312

BAD ORIGINAL

```

        jmp      _sarc_13          ;go to do the crc check

_sarc_00
;ISO new CRC_16 controls:

        move     x:crcbits,r1      ;get NEW bit ctr for span over CRC-16
        move     #5ffff00,x0      ;NEW: seed the checksum with F's

_sarc_13
        jsr      crc              ;do the checksum

;now, insert 16 bit checksum value

        move     y:<crcaddr,r0     ;address for start of the checksum
        move     a1,x:chksum        ;save checksum returned from crc run
        clr      a                  ;set up to shift checksum
        move     x:chksum,x1        ;set up to shift checksum
        move     x:chksum,a0        ;set checksum in lower part of reg

;isolate the bits to shift for storing:
; part in crcaddr and part in crcaddr + 1
; or all in crcaddr

        move     #>24,b            ;get bits in a word
        move     y:<crcsc,x0        ;get bit offset to store CRC checksum
        sub      x0,b #>NCRCBITS,x0 ;get bits remaining in word
        cmp      x0,b b1,y1        ; & get number of bits for CRC checksum
        ;test if CRC wholly in one word
        ; & save number of bits for 1st shift
        jeq      _no_shift         ;if equal, no shift
        jgt      _one_shift        ;if more than enough room in word

;we have to do two shifts for overlapping 2 words
; 1. shift the checksum over two bytes to position for shift into a1

        do       #24-NCRCBITS,_shift_a
        asl      a

_shift_a

; 2. shift bits to offset into a1

        do       y1,_store_1st
        asl      a

_store_1st

; 3. store 1st portion from checksum into 1st word

        move     a1,x1              ;bits for 1st word
        move     y:(r0),b           ;get 1st word at that address
        or       x1,b              ;set the low bits (were 0) to sum
        move     b1,y:(r0)-         ;store back into the frame
        ; & increment for 2nd word
        jmp      _shift_1          ;now store 2nd portion in 2nd word

_one_shift

;checksum fits within the 1st word

```

313

BAD,ORIGINAL



```
sub    x0,b           ;calculate numb bits to shift
do     b,_shift_1     ;shift up to offset for CRC
asi    a

_shift_1
;store shifted checksum value

move   a0,x1          ;a0 now positioned

_no_shift
;last NCRCBITS at that address

move   y:(r0),b        ;get the word at that address
or     x1,b            ;set the low 16 bits (were 0) to sum
move   b1,y:(r0)       ;store back into the frame
move   #-1,m0          ;restore to linear buffer control

rts
```





```

    opt    fo,mex
;
; 1994. Copyright Corporate Computer Systems, Inc. All rights reserved.
;
; \UXCODE\setctls.asm
;
; title    'Encoder set transmission line controls'
;
; This routine is used to interpret the transmission line selection
; and phase lock loop controls to set the variables required
; for the bit allocation conditions and output line selection
; and front panel leds.
;
; destroyed:
;     register a
;     register x0
;     register r0
;     register r1
;     register r2
;
; include 'def.asm'
; include 'box_ctl.asm'
;
; org      phe:
;
setctls
; initialize stereo control settings to reflect current transmission
;
    bclr    #SPLIT_MODE,y:<stereo
    bclr    #SPLIT_MONO_FRAME,y:<stereo
    bclr    #NO_LINES,y:<stereo
    bclr    #BOTH_LINES,y:<stereo
    bclr    #SUMMARY_ALARM,y:<stereo
;
; check the selected transmission lines and the phase lock loops
    move     #select1,r0          ;addr of the line 1 select flag
    move     #select2,r1          ;addr of the line 2 select flag
    jset     #0,x:(r0),_ctls_10   ;if line 1 selected
    jset     #0,x:(r1),_ctls_20   ;if line 2 selected
;
; neither line selected
    bset     #NO_LINES,y:<stereo
    jmp      _ctls_20
;
_ctls_10
; line 1 selected, check if line 2 also selected
; and if so, indicate both lines selected
    jclr     #0,x:(r1),_ctls_20   ;if line 2 not selected
;
; both lines selected, set as redundant
    bset     #BOTH_LINES,y:<stereo
;
_ctls_20

```

3/5

```
; if the device supports it (DUAL_LINES in box_ctl.asm is set to 1),
; check if redundant should be set to split mode and whether bit allocation
; should account for block sequence numbering:
;   yes if (128 or 112) bit rate qualifies (#SPLIT_APPLIES)
;   AND neither line is specifically selected

    if DUAL_LINES==1

        jclr    #SPLIT_APPLIES,y:<stereo,_ctls_40
        jclr    #NO_LINES,y:<stereo,_ctls_40
        bset    #SPLIT_MODE,y:<stereo

; further, if the receiver has a problem with one line,
; go into split MONO frame mode for frame 1.2 the normal size

        TST_CLR_REC_A_ERROR_CD,_ctls_40
        bset    #SPLIT_MONO_FRAME,y:<stereo

    endif

; indicate redundant mode, unless already set

    _ctls_40
    rts
```

316

BAD ORIGINAL

## 1. Introduction

### 1.1 cdqPRIMA Overview

The cdqPRIMA is an audio CODEC which is used to compress and decompress audio for transmission over a digital facility such as ISDN, T1, E1 and satellite. In addition to its audio compression capabilities, it has a rich set of monitor and control M&C features made possible by a powerful control processor and command language. These M&C capabilities provide the cdqPRIMA with unique capabilities not found in audio only CODEC's.

### *cdqPrima*™ Technical Features

<i>cdqPrima</i> Model	110	120	210	220	230
<b>Mechanical Features</b>					
Dimensions: 19" Rack Mount	1U high	1U high	2U high	2U high	2U high
Digital Interface Module slots	1	1	3	3	3
World Power Supply, rear power switch	X	X	X	X	X
Dial and control keypad	X	X	X	X	X
Backlit LCD display	character	character	character	character	graphic
Digital LED average & peak VU meters		X		X	X
L/R correlation & stereo image display		X		X	X
Scrolling text messages on VU meters		X		X	X
Intelligent headphone monitor system		X		X	X

X = always present • = hardware/software option; for example, • 3 means optional 3

<i>cdqPrima</i> Model	110	120	210	220	230
<b>Compression Algorithms</b>					
CCS MUSICAM	X	X	X	X	X
ISO/MPEG Layer II	X	X	X	X	X
CCITT G.722	X	X	X	X	X
16, 24, 32 & 48 kHz sampling rates	X	X	X	X	X
22.05 & 44.1 kHz sampling rates	•	•	•	•	•
Additional algorithm capacity	X	X	X	X	X

X = always present • = hardware/software option; for example, • 3 means optional 3

317

BAD ORIGINAL



<i>CDOPrima</i> Model	110	120	10	220	230
<b>Audio I/O, SMPTE &amp; Ancillary Data</b>					
18-bit A/D and D/A converters	X	X	X	X	X
Gold plated Neutrik® XLR audio connectors	X	X	X	X	X
AES/EBU, S/PDIF	• DB9	DB9	XLR	XLR	XLR
Automatic rate adaptation	X	X	X	X	X
Optical Digital I/O			•	•	•
Spectrum analyzer & phase display					X
SMPTE Time Code			•	•	•
Asynchronous ancillary data	X	X	X	X	X
Synchronous ancillary data	•	•	•	•	•

X = always present • = hardware/software option; for example, • 3 means optional 3

<i>CDOPrima</i> Model	110	120	210	220	230
<b>Command and Control</b>					
68020 Integrated Support Processor	X	X	X	X	X
Software update via RS232 & inband ISDN	X	X	X	X	X
J.52 (H.221) BONDING	X	X	X	X	X
Extensive on-line help	X	X	X	X	X
Headphone select and level control keypad		X		X	X
4-button cue keypad		X		X	X
Hot keys & extended feature keypad					X
Full remote control via RS232 & RS485	X	X	X	X	X
Front panel RS232 remote control port		X		X	X
Optically isolated remote control inputs	• 4	• 4	• 8	• 8	• 8
Dry floating relay contacts or TTL outputs	• 4	• 4	• 8	• 8	• 8
Virtual control lines connecting each unit	12	12	12	12	12
RS232 control port, no modem control	X	X			
RS232 control port, full modem control			X	X	X
RS485 control port			X	X	X
Programmable summary alarm relay	X	X	X	X	X
Programmable silence detector		X		X	X
Programmable peak level detector		X		X	X
Bit error rate detector	X	X	X	X	X
Out-of-frame detector	X	X	X	X	X

X = always present • = hardware/software option; for example, • 3 means optional 3

<i>cdqPrima</i> Model	110	120	210	220	230
<b>Additional Options Available</b>					
ISDN/X.21/RS422/V.35 DIF modules	• 1	• 1	• 3	• 3	• 3
Windows® remote control software	•	•	•	•	•
Psychoacoustic parameter adjustment	•	•	•	•	•
ITU-T J.52 error protection	•	•	•	•	•
Analog stereo input limiter	•	•	•	•	•

X = always present • = hardware/software option; for example, • 3 means optional 3

The cdqPRIMA family falls into two broad categories, the 1xx and the 2xx families. The 1xx family is 1U (1.75") high and holds 1 Digital Interface Module (DIM) while the 2xx family is 2U (3.5") high and holds 3 DIM's. Each DIM connects the cdqPRIMA to the digital transmission facility.

The block diagram of the cdqPRIMA is shown below.

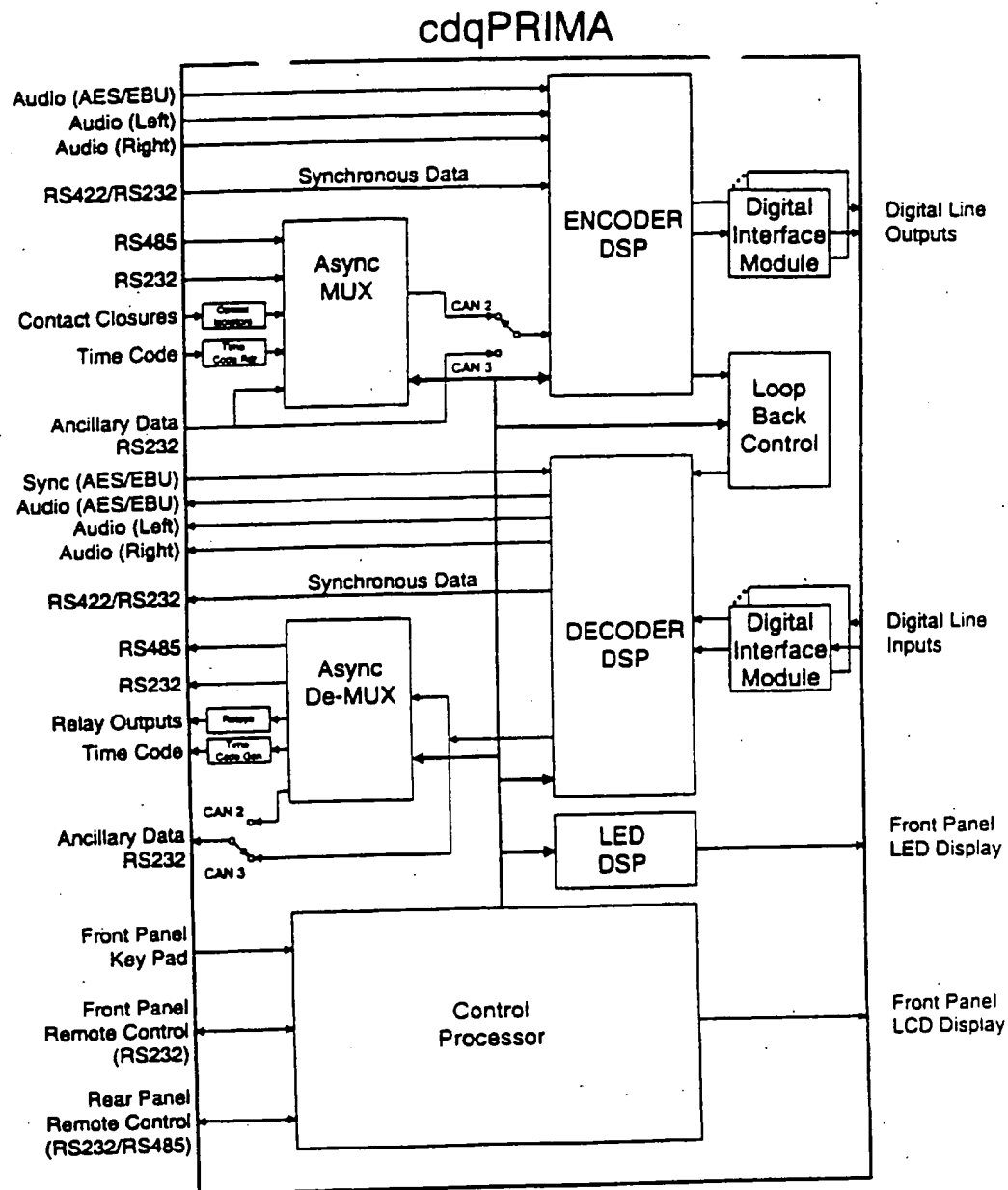


Figure 1  
PRIMA High Level Block Diagram

**Figure 1-1**  
cdqPRIMA High Level Block Diagram

### 1.1.1 Model 110 Description

The cdqPRIMA 110 provides a rich set of basic features which are included in the entire cdqPRIMA family such as a LCD display and control keypad in a 1U (1.75") 19" unit. The keypad allows operation of all the CODEC features. This unit also includes a rear panel remote control connector, 4 optical isolated monitor inputs and 4 control relay outputs. Ancillary data is also included in the cdqPRIMA 110. One Digital Interface Module (DIM) can be used to interface with digital networks.

### 1.1.2 Model 120 Description

The cdqPRIMA 120 builds on the model 110 by adding a LED level display, front panel headphone output, front panel remote control and AES/EBU digital audio I/O. The AES/EBU digital audio interface on the 1xx series utilizes a DB9 connector (an adaptor cable is available to convert from the DB9 to the standard XLR connectors). The keypad of the 120 includes buttons to control the headphone output source and level.

The LED level display provides a sophisticated level meter with peak hold, as well as stereo image and stereo correlation capabilities. The LED level display can also display scrolling messages to the user. Such messages are helpful in alerting and cueing.

The 120 adds additional keys to the keypad for headphone control.

### 1.1.3 Model 210 Description

The features of the model 210 are identical to the 110 with several additional features. The 2xx series is housed in a 2U (3.50") by 19 inch enclosure and includes 8 optical isolators and 8 relays. SMPTE time code and optical digital audio are optionally available. Three Digital Interface Modules (DIM's) can be used for interfacing to digital networks. On this model, the AES/EBU connectors are XLR instead of the DB9 on the 1xx series.

### 1.1.4 Model 220 Description

The features of the model 220 are identical to the 120 with the addition of 4 more optical isolators and 4 more relays.

### 1.1.5 Model 230 Description

This model provides all of the features of the 220 with the addition of a graphics display which can be used for measurements such as real time spectral analysis. The 230 also provides an enhanced keypad which adds measurement hot keys plus user programmable hot keys.



## 1.2 Digital Transmission Networks

### 1.2.1 Overview

The selection of the digital transmission facility must be considered when using the cdqPRIMA. The terrestrial network falls into two broad classification and these are the dedicated and switched networks. The dedicated network is, as the name implies, a dedicated path between two points. Examples of dedicated services are DDS56, T1 and E1. Typically, dedicated service is expensive but should be use if continuous connectivity is anticipated. If a dedicated or leased line is appropriate, it must have a CSU/DSU (Customer Service Unit / Data Service Unit ) installed at each end. These units are responsible for converting the V.35 or X.21 signals into signals compatible with the network. They are relatively inexpensive and readily available from numerous manufactures and require no special instructions.

The digital switched network is attractive when occasional use is required because the cost of the service is computed based on a monthly fee plus the actual time the service is used. This is exactly like a conventional phone and the rates charged by the service providers are relative inexpensive and comparable to standard telephone rates.

Two examples of switched long distance terrestrial networks are the ATT ACCUNET Switched 56 network/ISDN and the Sprint VPN network. Both are digital networks and are candidates for use with the cdqPRIMA. The Sprint VPN network uses digital lines which were designed for speech and includes digital echo cancelers. The effect of these echo cancelers is to modify the digital bitstream in an attempt to remove what it thinks are echoes. This modification of the digital bit stream is disastrous to the cdqPRIMA because it expects the receiver to receive a binary 1 when it transmits a 1. Fortunately, the echo cancelers are easily disabled by using a proper CSU/DSU. In particular, a CSU/DSU must be equipped with an echo canceller disabler if it is to be used in the Sprint VPN network. This is a common option in switched CSU/DSU's and must be ordered if the long distance carrier is Sprint.

The ATT Accunet Switched 56 network or ISDN is intended for data and voice and does not require echo suppression facilities in the CSU/DSU.

There is another consideration when using the terrestrial switch 56 kb network and that is 4 wire verses 2 wire. In various regions of the United States, different regional operating companies use different technology to transmit the 56 kb data from the customer premise to the central office. The two technologies are called 2-wire and 4-wire. When ordering the local phone line (local loop), you must inquire about the circuit type - 2 wire or 4 wire and then order an appropriate CSU/DSU.

Satellite facilities require no special attention. Only a standard 56 kbps, 64 kbps, ... 384 kbs data line is required.

The cdqPRIMA is relatively immune to digital bit errors. If a binary 1 is occasionally changed to a 0 or visa versa, it has minimal impact. Synchronization is maintained even during error burst of up to .1 second. However, in either the satellite and terrestrial facilities,

322

BAD ORIGINAL





a slip (the complete loss or addition of a bit) causes the receiver circuitry to lose lock and then require framing. This entire process is statistical but usually only takes about .2 seconds. During this time, the receiver mutes and no audio is output.

## 1.2.2 Digital Transmission Facilities

### 1.2.2.1 Digital Data Service ("Nailed-Up" DDS)

This is the original digital data service. It provides 56 kbs over a dedicated circuit. This technology is based on the telephone companies internal 64 kbs systems but 1 bit out of each 8 is robbed from the user for use by the telephone company to provide signalling information. This signalling information conveys such information such as dialing digits and on/off hook.

### 1.2.2.2 Switch 56

Switched 56 was the first switched digital technology transmission technology provided by the telephone companies. It utilizes the 56 kbs transport technology within the telco's as the DDS service described above.

### 1.2.2.3 The ISDN Basic Rate Interface (BRI)

ISDN is a new technology which is used to transport either 56 or 64 kbs. Utilizing ISDN, a single copper wire pair from the telephone company central office to the customer premises (a basic rate interface - BRI) can transport two B channels and one D channel. Each B channel can be either 56 or 64 kbs and the D channel transmits 16 kbs.

ISDN is computer to computer communication because it allows the central office computer to communicate with the customer premises computer. This customer premises computer is called a terminal adaptor (TA). This sophisticated computer to computer communication is accomplished over the D channel and does not rob any bits from either of the B channels. Since the central office computer is in contact with the customer premises computer, sophisticated communication is possible. For example, the central office computer can ask the customer premises computer if it will accept a data call at 64 kbs.

ISDN is the low bandwidth low cost interconnect method provided by the telephone companies. The rates of ISDN are similar to a normal analog telephone line.

### 1.2.2.4 Primary Rate ISDN (T1 & E1)

While ISDN provides 64 kbs service, T1 provides 24 64 kbs channels (1.544 mbs). E1 provides 32, 64 kbs channels. This increased bandwidth comes at an additional cost.

323

BAD ORIGINAL



#### 1.2.2.5 Long-Distance Interconnectivity (56-to-64kbps)

ISDN and switched 56 are available internationally. 56 kbs ISDN interconnects with switched 56 both nationally and internationally. This capability provides incredible world wide low cost connectivity.

### 1.2.3 Other Digital transmission Paths

While terrestrial facilities such as ISDN are popular, there are several other technologies for digital transmission which should be considered. RF transmission facilities form another class of transmission and are an alternative to terrestrial transmission.

#### 1.2.3.1 Spread-Spectrum

Spread spectrum RF transmission allows multiple transmitters to operate at the same frequency without interference. There is a practical limit to the number of transmitters which can be simultaneously transmitting but spread spectrum modulation is a useful method in light of the current US FCC regulations which allow low power transmitters in the 900 MHz frequency region.

Spread spectrum can be used for point to point or point to multipoint transmission. It is primarily used for point to point transmission.

Digital spread spectrum transmission communications systems are an excellent candidate for use with the cdqPRIMA

#### 1.2.3.2 Satellite Links

Satellite transmission is used primarily for point to multipoint transmission. Such systems are used to broadcast to many listeners.

The cdqPRIMA is perfectly suited to work in digital satellite systems.

## 1.3 Compression Algorithms

### 1.3.1 CCS MUSICAM Digital Audio Compression

#### 1.3.1.1 Introduction

Developments in the fields of consumer audio electronics and professional audio processing have been increasingly influenced by digital technology. Until five years ago, developments in the field of source coding were mainly restricted to the bit-reducing coding of speech signals for telecommunications applications.

Today, source coding techniques are playing an even greater role in the field of high quality digital audio. The reasons for this are the direct relationship between the low bit

324

BAD, ORIGINAL



rates associated with compression and the costs associated with the transmission and storage of compressed audio.

The bit-rate for high-quality stereo audio signals (1,411 kbs for a CD) can now be reduced by the MUSICAM algorithm to about 200 kbs. This is the result of major progress in the development of source coding techniques that utilize knowledge of the human ear. This means that the average quantization of the audio signal at a sampling rate of 44.1 kHz would be approximately 2 bits per sample in the mono channel instead of the 16 bits per sample used in CD's. Despite this high reduction in the bit rate, no quality differences are discernible to a trained ear. A slight impairment only becomes audible at higher compression rates. Additionally, MUSICAM offers the flexibility of independently adjustable audio sampling rates (32 kHz, 44.1 kHz, 48 kHz...) and digital bit rates (56 kbs, 64 kbs, 112 kbs, 128 kbs, 192 kbs, 256 kbs, 384 kbs...) as well as embedded data within the audio bit stream. All of these features are incorporated in the recently approved ISO MPEG audio standard. No other audio compression algorithm has undergone the scrutiny and testing subjected to MUSICAM as a result of the ISO selection process. The ISO standards committee has selected a truly universal digital audio source coding system with the flexibility to meet different system demands. Current and future audio systems adhering to the ISO MPEG audio standard will be able to interoperate easily and reliably. This will allow manufacturers to build sophisticated audio equipment and consumers to purchase hardware without the fear of obsolescence.

#### 1.3.1.2 MUSICAM Compression Concepts

The main principle of MUSICAM is the reduction of redundancy and irrelevance in the audio signal. Every audio signal contains irrelevant signal components that have nothing to do with the identification of the audio signal (i.e., determination of timbre and localization). These irrelevant signals are not significant to the human ear and are not required by the information processing centers in the brain. The reduction of irrelevance means that these signal components are not transmitted. This results in a lower bit rate without any perceived degradation of the audio signal. Furthermore, it is possible to allow a certain degree of quantizing noise that is inaudible to the human ear due to the masking effects of the audio itself. Every audio signal produces a masking threshold in the ear depending on a time varying function of the signal. To understand this masking effect, the concept a masking tone must be defined. A masking tone is simply a high amplitude audio signal occurring over a relatively narrow frequency span and is often called a masker. Typically, in an audio signal there exists a number of these masking tones occurring at several different frequencies.

A masking tone renders smaller amplitude tones close to it inaudible due to its masking effect. The exact shape of the masking effect is called the masking threshold. The aggregate of all the maskers defines a global masking threshold and the parts of an audio signal below the global masking threshold are inaudible. They are said to be masked and therefore need not be transmitted. Other signal components above the masking threshold only require the level of quantization to keep quantization noise below the masking threshold, and thus the quantization induced noise remains inaudible. Quantization noise

325

BAD ORIGINAL



can be better adapted to the masking threshold of the human ear by splitting the frequency spectrum into sub-bands.

The quantization of the analog time samples required for each sub-band is dependent on the minimum masking value in each sub-band. This minimum masking level is a measure of the allowed quantization noise that is just below the level of perceptibility. Sub-bands whose desired signals are well below the masking threshold (and are thus irrelevant for the human ear) do not need to be transmitted.

In each 24 millisecond period, a calculation of the masking threshold is performed for each sub-band. This threshold is then used to compute the psycho acoustically best allocation of the available bits. This process is called dynamic bit allocation. Audio data is quantized using the dynamic bit allocation and thus the required bit rate for time-variant audio signal's changes continuously due to the changing masking threshold. If there is an insufficient number of bits to hide the quantizing induced noise completely, then the noise is placed in the least objectionable place in the audio sample. If there is an excess number of bits, then the extra bits are used to reduce the quantizing induced noise to as low as possible level. The allocation of the extra bits is crucial and allows multiple encode-decode cycles as well as post production of the audio.

The total transmitted bit stream contains quantized audio values as well as auxiliary information describing bit allocation and scale factors, all of which are required by the decoder to reproduce the audio information.

The scale factors are determined by searching for the maximum sampling value in each sub-band and quantizing the result using 6-bit sampling. The scale factors have a dynamic range of 120 dB that is sufficient for future encoding for quantized PCM signals using up to 20-bit sampling yet still retain their dynamic range. All necessary information is encoded into MUSICAM frames each of which represents about 24 milliseconds of real-time audio.

All the complex calculations of the MUSICAM algorithm are performed by the encoder. Decoders are designed to be universal. MUSICAM decoders can be constructed which correctly decode and play back audio information that has been encoded by a range of MUSICAM encoders. This aspect of the MUSICAM algorithm is crucial because it enables refinements in the encoding process to further improve performance without impacting decoders that are already installed.

### 1.3.1.3 Performance Considerations

#### 1.3.1.3.1 Introduction

Before discussing the various quality aspects of MUSICAM, it is necessary to define the terms used to represent the field of use of the audio. The 4 commonly discussed fields of use are:

- Contribution

326

BAD ORIGINAL



- Distribution
- Emission
- Commentary

The term contribution grade is used to describe quality suitable for digital mastering. Its use would be in the transmission of a digital master from one archive to another. It is assumed that the original copy is in a 16 bit linear PCM format and it is to be compressed, transmitted, decompressed and stored in a 16 bit linear PCM format at the distant end. Because the audio is the source of future compression/decompression cycles, any contribution grade compression system must be able to withstand many encode-decode cycles and post production without any apparent degradation.

Distribution grade systems are used to transmit audio between two storage devices. However, the number of encode-decode cycles is limited to only a few. Distribution grade systems are used when the number of audio compression-decompression cycles is limited.

Emission grade systems are used when there is only one compression-expansion cycle anticipated. This is the case when audio is compressed and transmitted from one place to another, decompressed and stored on an analog tape and the only future manipulations done are in the analog domain.

Commentary grade systems are used for transmitting voice grade audio.

These definitions make no mention of the analog bandwidth or the exact definition. They are vague terms used to describe ability of the audio to withstand multiple encode-decode cycles. In all cases, the compressed audio is assumed to be indistinguishable from the original.

#### 1.3.1.3.2 ISO Background

The only independent measurements of audio quality of MUSICAM types of compression systems have been done by the MPEG ISO committee. Four algorithms in July of 1990 were tested and the winner according to the rules of the tests was MUSICAM. This algorithm was adopted and it was agreed that, to the extent possible, the best features of the second place algorithm, ASPEC, would be incorporated into MUSICAM to produce the final ISO standard.

The ISO committee decided to have a layered standard with 3 layers. Layer 1 is a very simplified version of the original MUSICAM algorithm. Layer 2 is essentially the MUSICAM algorithm as tested, and Layer 3 is a modification of Layer 2 that includes various features of ASPEC. It was anticipated that the resulting audio quality would improve with higher layer number. After the layers were defined, they were implemented according to the standard and each layer was tested in the May 1991 tests.

The results of these tests were surprising because Layer 3 scored lower than Layer 2. It has recently been decided that additional work on Layer 3 was needed and that layer

327

BAD ORIGINAL



would be retested in December of 1991. Layers 1 and 2 have been frozen in their present state because they have met their design objectives. As a result of the ISO effort, the MUSICAM algorithm is now properly called the MPEG Layer 2 compression algorithm.

It is clear from the most recent ISO tests that no compression scheme performs acceptably at 64 kbs. Work at that bit-rate is the subject of further research and will be addressed in a future standard.

The intensity or joint stereo mode of compression supported by Layer 2 (called Layer 2A) was not tested during the May 1991 tests. It is important to recognize that the ISO tests have provided a wealth of knowledge about the MPEG Layer 2 algorithm. Other algorithms such as SEDAT, AC-2 and APT-X did not even participate in the ISO tests and their strengths and weaknesses are unknown. It is certainly clear that MPEG Layer 2 has been demonstrated to be a superior algorithm. This claim can be supported by a large body of test data. Other algorithms have little or no independent test data to substantiate their quality claims.

#### 1.3.1.3.3 Quality vs. Bit Rate

The MUSICAM design allows the digital bit-rate, analog bandwidth and quality to be generally related by the formula

$$\text{Digital Bit-Rate} \times \text{Quality} = \frac{\text{Digital Bit-Rate}}{\text{Analog Bandwidth}}$$

As indicated above, the quality increases as the bit-rate increases and the analog bandwidth is kept constant. Similarly, if the digital bit-rate is kept constant, and the analog bandwidth is decreased, then the quality improves.

The ISO test in Stockholm in May 1991 has demonstrated that at a digital bit rate of 256 kbs per stereo channel; MPEG Layer 2 is statistically identical to the original signal. This means that the panel of approximately 60 highly trained listeners could not distinguish the original uncompressed source material from the audio compressed by the MPEG Layer 2 algorithm. The conclusion of the ISO tests (at 256 kbs per stereo channel) was that MPEG Layer 2 is transparent. MPEG Layer 2 scored 5 on the MOS (mean opinion score) scale where the lowest is 1 and the highest score is 5.

It is important to note that no other algorithm tested at ISO (including ASPEC) was considered transparent in the 256 kbs stereo tests. The ISO tests were conducted on stereo channels composed of two mono channels so that the combined bit rate was 256 kbs per stereo channel. The audio quality at 192 kbs was determined by ISO to be 4.5 on the MOS scale using stereo encoding and 2.0 for a mono channel at 64 kbs.

The MPEG Layer 2 algorithm provides the following qualities at various bit rates.

- contribution 384 kbs (stereo, Layer 2)
- distribution 256 kbs (stereo, Layer 2)

328

BAD ORIGINAL



- emission 192 kbs (stereo, Layer 2A)
- commentary 64 kbs (mono, Layer 2)

The classification of 192 kbs for the emission grade is based on recent work at the IRT (Institute für Rundfunk-Technique) and relies on the intensity (joint) stereo coding technique for additional compression.

#### 1.3.1.4 Tolerance to Transmission Errors

The ISO MPEG Layer 2 data block consists of two parts. The first is the header and consists of framing, bit allocation, scale factors and other side information. The second part of the frame is the audio data. In the case of 256 kbs per stereo channel, the length of a 24 millisecond frame is 6144 bits, the header part of the frame is approximately 300 bits and the remainder of the frame is the audio data. The bit integrity of the entire header is vital since it defines the layout of the remainder of the frame. Any bit error in the header causes degradation because the following parts of the frame would be decoded incorrectly and thus 24 milliseconds of audio would be lost.

An error in the data part of the frame can range from imperceptible to just barely noticeable. This is because a single bit error only affects a single data sample and thus only a very small time. If the bit error occurs in the least significant bit of the data sample, the effect of the error is minimal. However, if the error occurs in the most significant bit (the sign-bit) then the effect is more pronounced.

The header of an MPEG frame is protected by an error protection polynomial and provides the ability to detect errors that occur in the header. The data part of the frame is unprotected and any error occurring in the data part of the frame remains. The error strategy used for the ISO MPEG system is as follows. If an error is detected in the header, the last frame (24 milliseconds) of audio is repeated. If, in the succeeding frame, an error is detected in the header, the second and all succeeding frames with errors are muted. This error mitigation technique has been shown to be effective for bit rates of approximately 10-5. This error rate represents error rates easily achievable by transmission systems. Using this strategy, there is a smooth degradation of the audio quality as the error rate increases until the error rate becomes excessive at this point the audio output mutes.

#### 1.3.1.5 Tolerance to Multiple Processing

To understand the effect of multiple encode and decode cycles it is important to review the predominant effect that allows MPEG audio to achieve its compression. This is the hiding of quantization noise under a loud signal. MPEG audio adjusts the degree of quantization induced noise in each sub-band and thus hides more noise (uses fewer bits) in the sub-bands that contain large amounts of audio energy.

The quantizing noise raises with each encode and decode cycle and after a sufficient number of cycles, the noise level becomes perceptible. The degradation process is gradual

329

BAD ORIGINAL



and depends upon level of the quantizing noise on the original. For example the following table list the approximate numbers of total encode and decode cycles before the noise

<i>Bit Rate</i>	<i>Number of</i>
384 kbs	15
256 kbs	5
192 kbs	2
128 kbs	1

**Table 1-1**

Number of transcodings vs bit rate

becomes significant.

It is important to understand that these are approximate and the exact number depends highly on the source material.

#### 1.3.1.6 Post Production Processing Effects

Post production processing of compressed audio is a complicated effect to model. For example, an equalizer changes the level of a range of frequencies, while limiting and compression are non-linear processes. Very little test data is available to ascertain the effects of post processing. Private communications with the IRT suggest that MPEG layer 2 is robust against the effects of post processing and the degree of robustness depends on the compression rate. In particular, 384 kbs audio is unaffected by post processing while 128 kbs audio is somewhat sensitive to post processing. It is not easy to define tests to measure the effects of post processing but an international standards body (CCIR) is specifically designing test to determine the effects of both transcoding and post processing. These tests were conducted in November of 1991 and represented the first time such tests were performed by an independent organization.

MPEG audio represents the most tested, documented and reviewed audio compression algorithm in the world. It is significant to note that no other compression technique has survived this crucial review process as well as the MPEG algorithm and, many other algorithms have elected not to participate in this review process. It is precisely these untested algorithms that make the boldest claims. MPEG audio provides the security of the international review process to insure the highest quality audio possible with today's technology.

330

BAD ORIGINAL





### 1.3.1.7 The MUSICAM Advantage

The MUSICAM digital audio compression algorithm has been designed to take advantage of future advances in psycho acoustic research. To make this possible, the decoder is designed to be a slave to the encoder. This technique allows the entire system to be upgraded by simply changing the encoder software. Once this change is made, the entire network is upgraded and the encoder enhancements are reflected at the output of all decoders.

The MUSICAM algorithm is designed to operate at multiple bit-rates. This gives the user the ultimate flexibility to make the tradeoff between quality and cost. The use of higher bit-rates (384 kbs) allows nearly an arbitrary number of transcodings and extensive post processing while still maintaining transparency. The middle bit-rates (256-192 kbs) allow lesser amounts of manipulation while the lower bit's rates (128 kbs) are the most sensitive the these effects. As advances in the research progress, today's bit-rates required to achieve a desired quality will decrease and the ease of MUSICAM to accommodate these advances provides a significant advantage. This is being demonstrated by the research into intensity coding of stereo signals. This shows that the data rate of 192 kbs for stereo signals will most likely be the new standard rate for transparent audio and will supplant the 256 kbs rate accepted as the standard today.

MUSICAM is able to embed other information within the audio bit stream. Again, in the MUSICAM design, the data rate of this ancillary information is completely flexible and thus is entirely in the hands of the system designer. This data rate is completely determined by the encoder and thus the may be changed at any time with no modifications to the decoders. The inclusion of data in the audio bit stream reduces the bits available for audio data and thus the system designer can make the delicate tradeoff between the ancillary data rate and audio quality.

The flexibility of MUSICAM to adapt to current and future needs is a powerful feature necessary to prevent the obsolescence of any system based on it. There is now no need to divine future system needs because the system can be easily be changed to accommodate its ever changing requirements.

#### Ancillary Data Port

The CDQPRIMA provides for transmission of asynchronous data via a RS-232 interface. This interface provides a transparent channel for the transmission of 8 data bits. The data format is 1 start bit, 8 data bits, 1 stop bit and no parity bits. This interface is capable of transmitting at the maximum data rate selected by the encoder and decoder data rate dip switches and thus no data pacing such as XON/XOFF or CTS/RTS is provided. Appendix C describes the encoder and decoder dip switches.

The encoder RS-232 data rate can be set from 300 to 19,200 bps. The use of the ancillary data channel decreases the number of bits available to the audio channel. The reduction of the audio bits only occurs if ancillary data is actually present. The data rate can be thought of as a maximum data rate and if there is no ancillary data present, then no data bits are transmitted. A typical example of this situation occurs when the CDQPRIMA



encoder is connected to a terminal; when the user types a character the character is sent to the decoder at the bit rate specified.

The setting of the decoder baud rate selection dip switches must be done considering the setting of the encoder. The decoder dip switches must be an equal or higher baud rate relative to the encoder. For example, it is possible to set the decoder ancillary baud rate to 9,600 baud. In this case, the encoder baud rate may be set to any value from 300 to 9,600 but not 19,200. If the decoder baud rate is set to a higher rate than the encoder, the data will burst out at the decoder's baud rate. The maximum sustained baud rate is controlled by the encoder.

The algorithm for the transmission of ancillary data is for the encoder to look during each 24 millisecond MUSICAM frame interval and see if any ancillary data is in its input buffer. If there are characters in the encoder's input buffer, then the maximum number of characters consistent with the selected baud rate are sent. During a 24 millisecond period,

<i>Bit Rate</i>	<i>Number of Characters</i>
300	1
1200	3
2400	6
3600	9
4800	12
7200	18
9600	24
19200	47

**Table 1-2**

Number of characters/frame (48 kHz)

the table below shows the maximum number of characters sent for each baud rate.

The CDQPRIMA provides no error detection or correction for the ancillary data. The user assumes the responsibility for the error control strategy of this data. For example, at an error rate of  $10^{-5}$  (which is relatively high) and an ancillary data rate of 1200 baud, 1 out of every 83 characters will be received in error. Standard computer data communication protocol techniques can be used to maintain data integrity.

When designing an error protection strategy, it must be remembered that the CDQPRIMA may occasionally repeat the last 24 milliseconds of audio under certain error conditions. The effect on the audio is nearly imperceptible. However, the ancillary data is not repeated.

### 1.3.1.8 Compatibility with older CCS CODECs

#### 1.3.1.8.1 CCS Old

See dave brown for an explanation.

#### 1.3.1.8.2 CCS New

See dave brown for an explanation.

### 1.3.2 Layer 3

ISO MPEG Layer 3 was an attempt of the ISO committee to utilize the best features of the algorithm which lost the ISO competition (ASPECT) with the winning algorithm (MUSICAM). The resulting algorithm utilizes the sub-band filter bank of MUSICAM with MDCT within each sub-band. The results of ISO and CCIR testing have shown that Layer 3 provides a small advantage only at 64 kbs mono and has the distinct disadvantage when cascaded. It is an extremely complicated algorithm and provides limited improvement at best.

### 1.3.3

The CdqPRIMA uses Adaptive Differential Pulse Code Modulation (ADPCM) to reduce the digital bit rate needed to transmit the digital representation of an analog signal. The CdqPRIMA digitizes the incoming analog signal with a 16 bit linear Analog to Digital converter (AD) 16,000 times per second. The Nyquist theorem states that at this sampling rate, an analog signal of up to 8,000 Hertz can be reconstructed from the sampled signal. Using this sampling rate and AD converter resolution, the following uncompressed bit rate is derived:

$$\text{PCM bit rate} = 16,000 * 16$$

$$\text{PCM bit rate} = 256,000 \text{ bits per second}$$

The cdqPRIMA then compresses this bit rate down to 64,000 or 56,000 bits per second using ADPCM.

To accomplish this compression, ADPCM utilizes the fact that the next sample of speech can be predicted by previous speech samples. The CdqPRIMA only transmits the difference between the predicted and actual sample. If the prediction process is effective, then the information to transmit consists of significantly fewer bits than the digital representation of the actual sample. The prediction accuracy is greatly enhanced by splitting the 8 kHz band into two 4 kHz bands. The signal in each band is predicted separately. This allows a more faithful representation of the analog signal than is possible by considering the whole 8 kHz band at once.

In conventional PCM, the binary representation of each sampled analog point is used. Differential PCM (DPCM) transmits the difference between the previous point and the



current point. In this scheme, the prediction process only involves the previous point. In fact, the predicted value of the current point is exactly the last point. In CCITT G.722 implementation of ADPCM, the predictor is very sophisticated and uses the previous 6 points to predict the current point. This results in a very accurate prediction and hence a very low bit rate.

#### **1.3.4 Future Algorithms & Prima Upgrade Capacity**

The cdqPRIMA has the capability to hold several audio compression algorithms. This permits the cdqPRIMA to be resistant to obsolescence. The cdqPRIMA can be downloaded from ISDN and thus the future upgrades are simple and effortless to install. This should be contrasted to the ROM type of update procedure currently employed by most CODEC manufacturers.

334

BAD ORIGINAL 

## **2. Installation**

### **2.1 Unpacking & Inspection**

Upon opening the shipping container, examine the cdqPRIMA for mechanical defects. Report any problems promptly to CCS. Plug the unit into the main power and turn on the unit via the rear panel power switch. The front panel LCD's should illuminate and display the power up sequence on the front panel LCD display.

### **2.2 Location of Units**

The cdqPRIMA has been designed to allow installation at locations with high RF fields.

#### **2.2.1 Environmental Considerations**

It is important that the ambient temperature specifications are met. It is usually possible to stack the cdqPRIMA units directly on top of other electronic equipment. It is important that the cdqPRIMA not be exposed to condensing humidity or fungal environments..

#### **2.2.2 Configuration Dependencies**

The cdqPRIMA can be used with a variety of digital transmission facilities. Typical applications consist of ISDN, satellite and dedicated facilities. The cable lengths for the interconnections can be from centimeters to kilometers. It is important to utilize twisted pair cable with an overall shield for the compressed audio interface. Flat ribbon cable should be avoided!

The digital audio interconnections are much less tolerant to longer cable lengths. Distances of 30 meters should be considered as an upper bound. Good cable construction is a necessity for the digital audio cables.

#### **2.2.3 Remote Control Considerations**

The cdqPRIMA is designed to be completely controlled remotely by a host computer. A rich command set can be used to control the entire operation of the cdqPRIMA. The section entitled cdqPRIMA Remote Control Commands contains a detailed description of all the remote control commands.

### **2.3 Connection to Network**

The cdqPRIMA family provides a variety of digital interfaces. Including V.35, X.21 leased circuit and RS422. Each of these digital interfaces requires clock and data to be exchanged between the cdqPRIMA and the terminal equipment. The cdqPRIMA always expects the clock to be provided by the terminal equipment. The encoder section outputs data synchronized with the clock and the decoder expects the data to be synchronized

335

BAD ORIGINAL



with the clock. Figures 5 and 6 show the interconnection of the cdqPRIMA to a generic piece of terminal equipment. The timing relationships are shown in Appendix B.

The data and clock lines are differential requiring a pair of wires for each signal. The control lines in the V.35 interface are single ended and require only one wire for each signal. The X.21 control lines are differential. The RS422 interface does not support any control lines. Any input control lines defined are ignored by the cdqPRIMA and any output control lines defined are held at constant values. See Appendix A for the

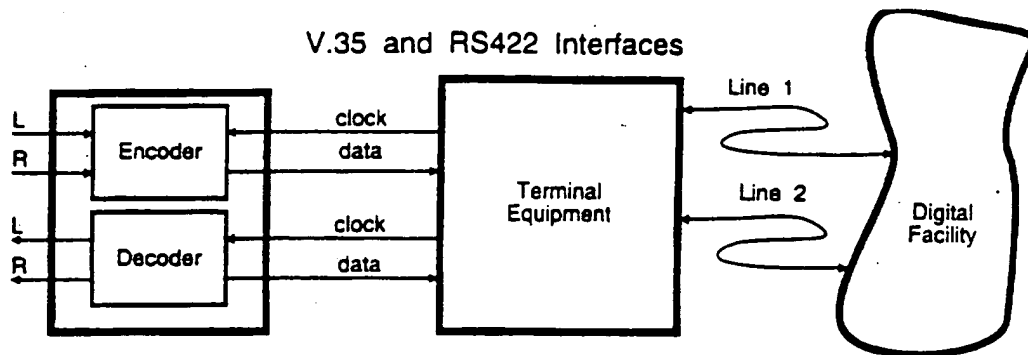


Figure 5  
Basic interconnection to digital network

Figure 2-1

Basic interconnection to digital network - RS422/V.35

definition the pins used for each type of interface.

Each interface defines a voltage level for each of the signals. In the case of V.35 and X.21, a connector type is also defined. The connector defined in the V.35 specification is not used by the cdqPRIMA because of its size. Instead, a smaller DB25 connector is used. In the case of the V.35 interface, the cdqPRIMA conforms to the electrical specification but requires an adapter cable to convert the DB25 connector to the connector specified in the V.35 specification. The connector and the pin-out chosen for the V.35 interface in the CDQPRIMA are a common deviant found in many systems. It is important to remember that V.35 has a separate clock for transmitted and received data. Appendix E describes the pin-out required for a DB25 to V.35 connector. The RS422 interface specification only defines the electrical voltages at the interface and leaves the pin-out and meaning of the pins to the hardware designer. The RS449 interface specification utilizes the electrical specifications of RS422 but specifies a mechanical connector. RS449 also specifies numerous control signals besides clock and data. The cdqPRIMA RS422 interface pin-out is specified in Appendix A. The RS422 interface also has a separate clock for the transmitted and received data. The cdqPRIMA RS422

interface also echoes the transmitter clock. If the terminal equipment clocks the encoder data with the echoed clock, then the cdqPRIMA may be located up to 4000 feet from the terminal equipment without having to worry about the encoder to clock skew.

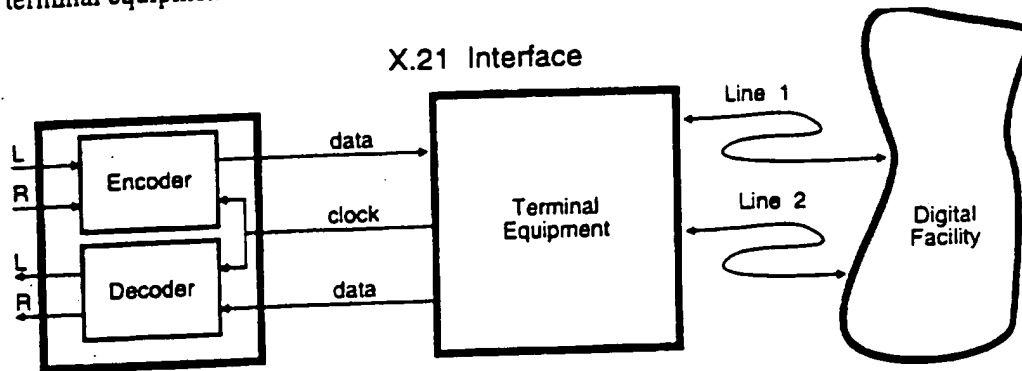


Figure 6  
Basic interconnection to digital network

Figure 2-2

Basic interconnection to digital network - X.21

The X.21 interface specification is in general a very complex specification. The general specification allows a mechanism for communication between the customer equipment and the network. This communication path can be used for things such as dialing. A sub-set of the specification, called the leased circuit, restricts the interconnection to only clock and data and a very simple control signal. The mechanical connector required is the DB15 with the pin-out specified in Appendix A. The electrical specification is RS422. The X.21 interface has only one clock for both the transmit and received signals.

Since the X.21 utilizes the RS422 electrical interface, the cdqPRIMA can use the same connector for both interfaces. In the case of the X.21 interface, the single clock is used internally for both the transmit and received timing. The selection of the type of digital interface is governed by rear panel dip switches. See Appendices C and D for the appropriate settings.

### 2.3.1 ISDN Card

The ISDN interfaces provide the following capabilities

- TA101 1 BRI S/T interface
- TA201 1 BRI S/T interface
- TA202 2 BRI S/T interface

337

BAD ORIGINAL



- TA211 1 BRI U interface (US only)
- TA222 2 BRI U interface (US only)
- The TA101 provides basic ISDN TA functions and requires a separate ROM for each country

### 2.3.2 Interface

The V.35 interface used on the cdqPRIMA utilizes the standard voltage levels and signals of the V.35 standard. It utilizes DB15 connectors instead of the standard large multipin connector. A cable adaptor is available which adapts the DB15 to the standard connector.

### 2.3.3 Interface

The X.21 interface provides the voltage levels, pinout and connector specified in the X.21 specification.

### 2.3.4 RS422 Interface

The RS422 interface utilizes the same DB15 connectors and voltage levels as used in the X.21 interface. It replaces the X.21 Control and Indicator signals with other timing signals.

## 2.4 Rear Panel Connectors

### 2.4.1 Analog I/O

The cdqPRIMA provides 18 bit AD and DA converters for the analog conversion modules. The analog sections of the cdqPRIMA are set to +18 dBu maximum input levels. Other analog input and output levels are possible by consulting CCS..

### 2.4.2 AES/EBU I/O

The AES/EBU digital audio interface standard provides a method to directly input (and output) audio information. This standard allows interconnection of equipment without the need for Analog/Digital conversions. It is always desirable to reduce the number of AD conversions since each time the conversion is performed, noise is generated. The cdqPRIMA allows digital audio input and output via a rear panel connector.

The cdqPRIMA model 1xx series, the AES/EBU connector is a DB9 due to space considerations. The cable drawing for an adaptor from the DB9 to standard XLR connectors is provided in the section labeled CABLE DRAWINGS.

The cdqPRIMA 2xx series uses the standard XLR connectors.





The AES/EBU digital input is rate adapted on input as well as output to eliminate any digital clock problems. The AES/EBU digital output from the decoder can be synchronized to a studio clock via an external AES/EBU sync input located in the rear of the cdqPRIMA.

Because of the rate adaptors, the input/output digital rates are not required to be the same as the internal rates. For example, it is possible to input 44.1 kHz AES/EBU digital audio input and ask the cdqPRIMA to perform compression at 48, 44.1 or 32 kHz (by using the front panel LCD display or the remote control **ESR** command). This is possible because the digital audio rate adapters.

Digital audio input sources can only be 32, 44.1 or 48 kHz. These input sampling rates are automatically sensed and rate adapted.

The compression algorithm at the encoder determines the digital sampling rate at the decoder. Thus the **ESR** command sets the internal sampling rate at the decoder. The AES/EBU digital output signal at the decoder is determined by the **DDO** command and can be a variety of values. See the **DDO** command for a detailed description.

The encoder receives direct digital input via the connector on the rear panel. Analog or digital (but not both simultaneously) signals may be input to the cdqPRIMA as selected by the front panel switch. If the digital input is selected, the CDQPRIMA locks to the incoming AES/EBU input and displays the lock condition via a front panel LED (not available on all models). If digital audio input is selected, the AES PLL lock light must be illuminated before audio is accepted for encoding. In normal operation, the CDQPRIMA locks its internal clocks to the clock of the telephone network. For loopback, it locks its clocks to an internal clock. In either case, the clock used by the CDQPRIMA is not at precisely the same frequency as the AES/EBU input. To prevent slips from occurring due the presence of two master clocks, a rate synchronizer is built into the encoder section to perform the necessary rate conversion between the two clocks.

The decoder outputs direct digital signals via the rear panel connector. Additionally, the decoder may be synchronized to an external clock by an additional connector (SYNC) on the rear panel. If no input is present on the decoder AES/EBU SYNC input line, then the output AES/EBU digital audio is generated by the internal clock source that is either at the telephone or internal clock rate. If the SYNC input is present, then the digital audio output is generated at the frequency of the SYNC input. The presence of a valid sync source is indicated by the illumination of the front panel AES PLL LED. The sync frequency may be slightly different from that of the CDQPRIMA clock source and again rate synchronism is performed to prevent any undesired slips in the digital audio output. The SYNC input is assumed to be an AES/EBU signal with or without data present. The CDQPRIMA only uses the framing for the frequency and sync determination.

#### 2.4.3 Power & Power Switch

This switch is used to control the main power to the cdqPRIMA.



#### **2.4.4 Remote Control**

This I/O port on the cdqPRIMA provides for either RS232 or RS485 remote control. It has the same capabilities as the front panel remote control. The choice of the RS232 or RS485 interface can be made by a remote control command or a front panel LCD command. A detailed description of the remote control commands is given in section entitled A Summary of cdqPRIMA Remote Control Commands.

#### **2.4.5 Ancillary Data**

The Ancillary Data connector provides an RS232 bi-directional interface for the transmission of asynchronous data. The data rates range from 300 to 38400 baud.

#### **2.4.6 Alarm**

This is a DPDT relay output whose function is controlled by the RLS action. See the section entitled cdqPRIMA Logic Language. It is often used as a summary alarm output to indicate the failure any major subsystem in the cdqPRIMA.

#### **2.4.7 1xx Series**

##### **2.4.7.1 Opto/Relay I/O and Sync Data**

For space reasons, the 4 optical isolated inputs, 4 relay outputs and the synchronous ancillary data I/O has been combined into one connector.

#### **2.4.8 2xx Series**

##### **2.4.8.1 Optical)**

The cdqPRIMA (on the 2xx models) provides an optional optical digital audio interface. This interface utilizes the EIA-J optical connectors. The functions of the EIA-J optical inputs are identical to the AES/EBU digital input connectors described above. The EIA-J connectors are enabled by a rear panel slide switch.

##### **2.4.8.2 Time Code**

The cdqPRIMA allows the transmission of timecode at rates of 24, 25, 29 and 30 frames per second. The cdqPRIMA automatically detects the presence of timecode at the encoder, converts it into a digital form and then multiplexes it into the ancillary data stream for transmission with the audio. At the decoder side, the ancillary data is separated from the audio and then demultiplexed. The time code is reconstructed

##### **2.4.8.3 Opto Inputs**

The optically isolated inputs on the 2xx series are identical to that of the 1xx series except that there 8 input sources.

340

BAD ORIGINAL

#### 2.4.8.4 Relay Outputs

The relay outputs on the 2xx series are identical to that of the 1xx series except that there are 8 input relays.

#### 2.4.8.5 Sync Data

The synchronous data port on the 2xx series is similar to sync data port on the 1xx series except that the output can be RS232 as well as RS484.

#### 2.4.8.6 RS232

The RS232 I/O connector is used to provide an additional port into the data multiplexor. It can be thought of as a second RS232 ancillary data port.

#### 2.4.8.7 RS485

The RS485 I/O connector is used to provide an additional port into the data multiplexor. It is a dedicated RS485 port and can be used to control RS485 equipment.

### 3. Feature Summary

#### 3.1 Async Ancillary data

##### Associated Remote Control Commands

CAN	Set ancillary data mode
CMA	Set MUX ancillary data baud rate
CDR	Set ancillary data rate for encoder and decoder DSP
DSB	Set decoder synchronous ancillary data bit rate
ESB	Set encoder synchronous ancillary data bit rate

The ISO-MPEG audio packet consists of the following parts:

- Header
- Audio Data
- Ancillary Data

If the sampling rate is 48 kHz, then the length of each packet is 24 milliseconds. The header consists of a 12 bit framing pattern, followed by various bits which indicate the data rate, sampling rate, emphasis, copyright, original .... These header bits are protected by an optional 16 bit CRC.

The Header is followed by the audio data which describes the compressed audio signal.

Any remaining bits in the packet are considered ancillary data. The format of the ancillary data is user defined. CCS has defined two ways of using the ancillary data. The first method has been used in the CDQ20xx series products and treats the entire data stream as one logical (and physical) stream of data.

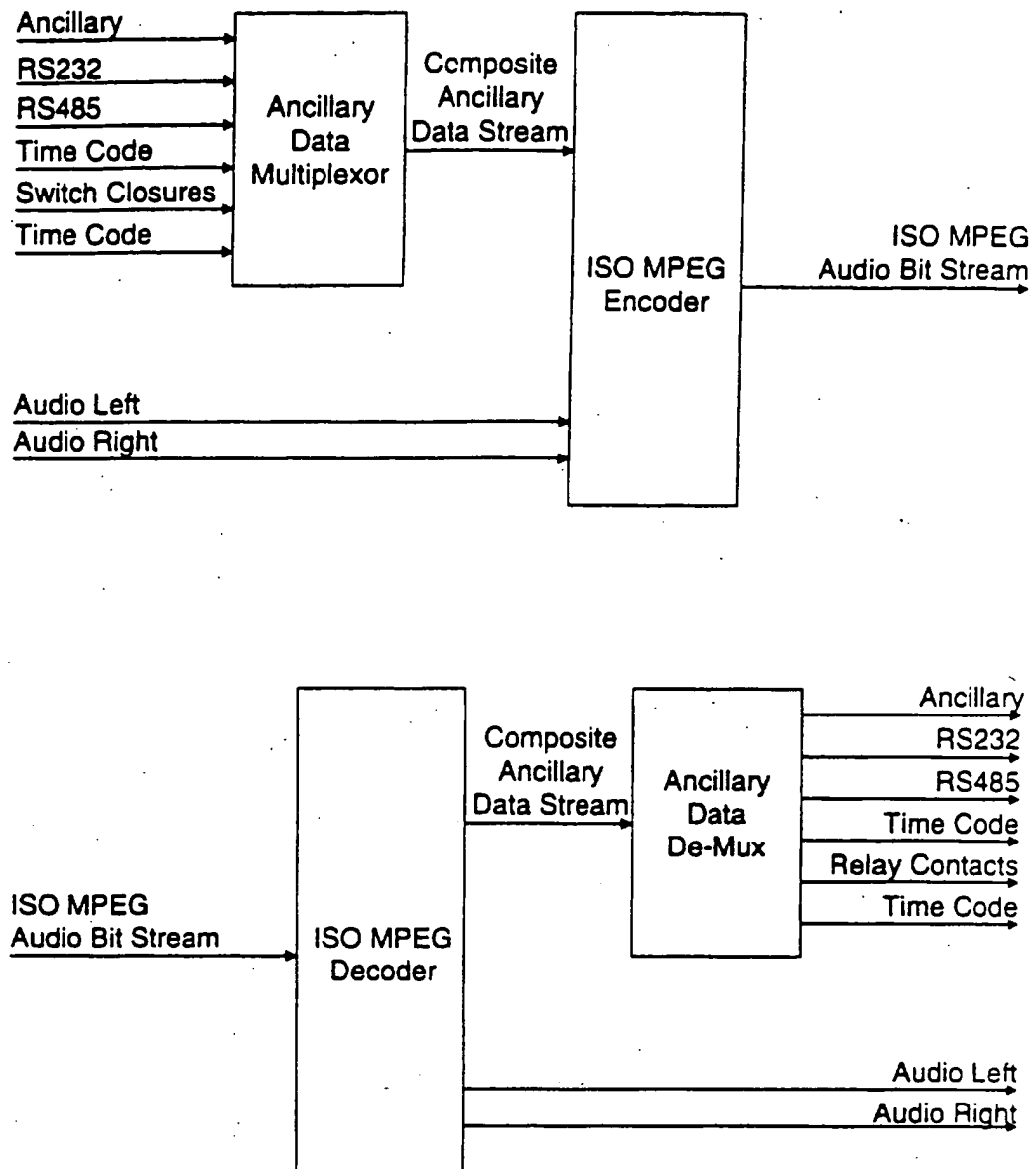
The cdqPRIMA series supports the older CDQ20xx ancillary data format as well as the newer cdqPRIMA format. This newer format allows the multiplexing of various logical and diverse data streams into one physical data stream. For example, switch closure, RS232 and time-code data are all multiplexed into a single physical data stream and placed in the ancillary data stream of the ISO MPEG packet.

The data rate from the Ancillary Data Multiplexor to the Encoder (and from the Decoder to the Ancillary Data Demultiplexor) is set by the **CDR** command. The data rate from the Ancillary connector into the Ancillary Data Multiplexor (and from the Ancillary Data Demultiplexor) is set by the **CMA** command. If **CAN** mode 2 is in use, then the **CMA** command has no meaning since the ancillary data is routed directly to and from the DSP's.

342

BAD ORIGINAL





**Figure 0-1**  
**cdqPRIMA ancillary data overview**

The synchronous ancillary data rates are controlled by the **DSB** and **ESB** commands.

The communication between the MUX and the encoder DSP and the DE-MUX and the decoder DSP is via an asynchronous communications channel. The data rate of both of these channels are simultaneously set by the **CDR** command

The RS232 ancillary data port can be used in several ways. It can be connected through the MUX/DE-MUX as described above or it can be connected directly to the encoder and decoder DSP's. Connecting directly to the encoder and/or decoder DSP's allows the highest baud rate (38,400) to be used but remove many useful features of the MUX. The output of the MUX may be connected directly to the DE-MUX and bypass the encoder and decoder DSP. This configuration is useful for testing.

<i>mode</i>	<i>description</i>
0	Direct connect - encoder DSP only
1	Direct connect - decoder DSP only
2	Normal mux mode
3	Direct connect - input to encoder DSP and output to decoder DSP (old CDQ20xx mode)
4	Input to MUX and direct output from decoder DSP
5	Direct input to encoder DSP and decoder DSP output to DE-MUX
6	Normal mux mode - DSP bypass

**Table 0-1**

Summary of **CAN** modes

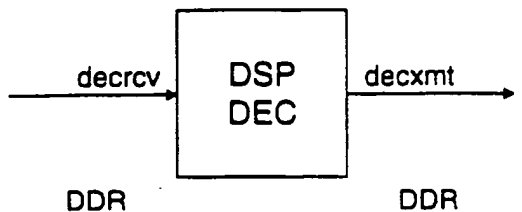
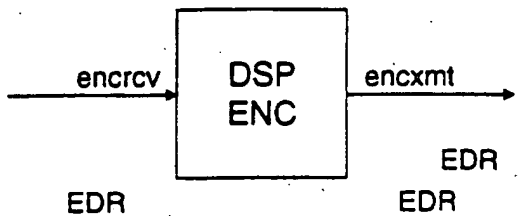
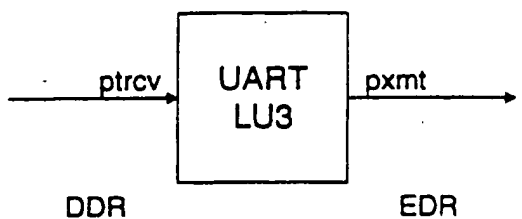
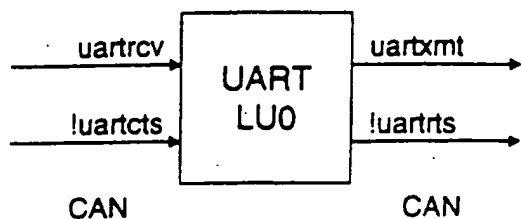
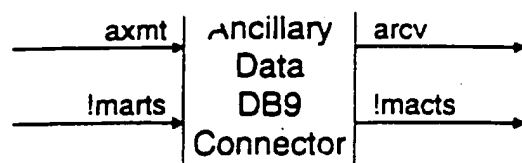
Mode 2 is the normal mode of operation when the data multiplexor is desired. Mode 3 bypasses the data multiplexor and connects the data at the Ancillary connector directly to the encoder and decoder DSP's. Mode 6 is useful for testing since it connects the multiplexor directly to the demultiplexor and thus bypasses the encoder and decoder DSP's.

### 3.1.1 Asynchronous ancillary data configurations

These various configurations are shown below.

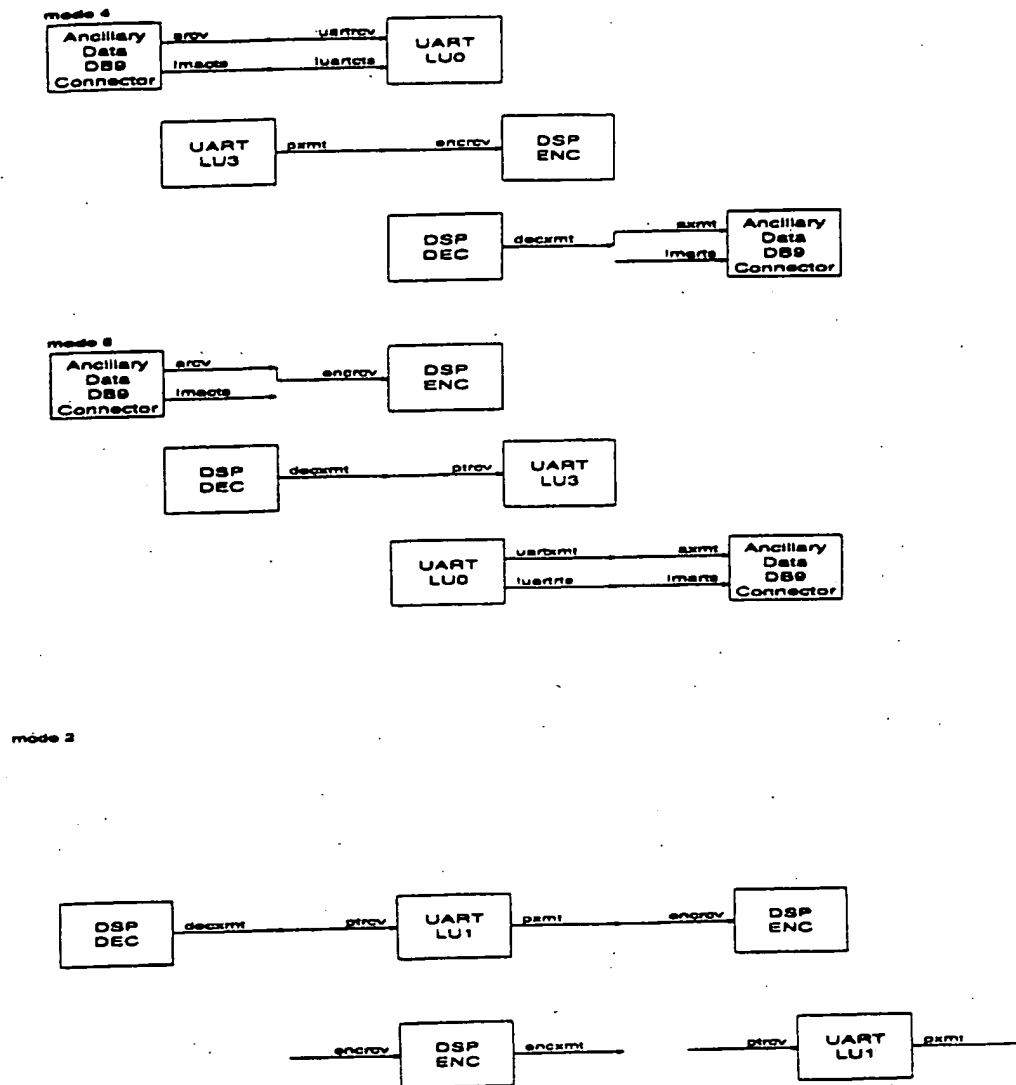
344

BAD ORIGINAL 



Commands to set the  
various baud rates  
EDR  
DDR  
CAN

Figure 3-2



**Figure 3-3**  
cdqPRIMA ancillary data switch configurations



mode 6

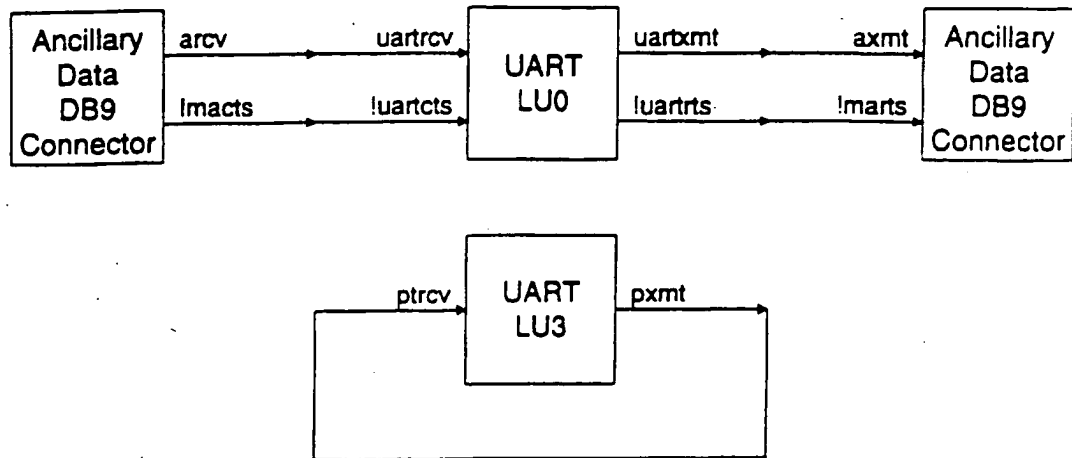


Figure 3-4

### 3.2 The Bit Error Rate Detector

#### Associated Remote Control Commands

MBC	Display BER counter
MBD	Set BER down count rate
MBL	Set BER count rate limit
MBR	Reset BER counter
MBU	Set BER up count rate

The bit error rate detector provides a method of monitoring the number of bit errors in the digital transmission path. The bit error rate detector is used when ISO frame protection is enabled. When each ISO/MPEG frame is received (every 24 milliseconds for 48 kHz sampling), the header CRC is checked for validity. If the frame header has a valid CRC, then the BER counter is incremented by a BER up count (any number from 0 to 9). If the frame is invalid, then the BER counter is decremented by the BER down count (any number from 0 to 9).

If the BER up count is set to 1 (by the **MBU** command) and the down count is set to 0 (by the **MBD** command), then the BER counter counts the total number of frames in error. If the up counter is set to 2 and the down count is set to 1, then the BER counter is sensitive to burst errors but not random errors.

The BER counter is compared to the BER threshold (set by the **MBL** command) to see if the counter is above or below the threshold. Actions such as closing a relay, dialing a phonenumber or lighting a LED or displaying a scrolling message can be taken.

The BER counter can be reset to 0 by the **MBR** command.

The current contents of the BER counter can be displayed by the **MBC** command.

### 3.3 Decoder

#### Associated Remote Control Commands

DAL	Set decoder algorithm
DBR	Set decoder bit rate
DCO	Set decoder decoding mode
DCS	Set channel copy/swap mode
DDA	Calibrate the DA converter
DIN	Set decoder - encoder interaction
DLI	Set decoder digital line format
DMD	Set decoder maintenance diagnostic mode
DMU	Mute decoder output channels
DSP	Scale factor protection
DRS	Print real-time decoder status bits

The decoder may be operated independently from the encoder by the proper setting of the **DIN** command. This can be extremely useful if the cdqPRIMA decoder is operated in a

348

BAD ORIGINAL



stand alone mode and is not controlled by the encoder. This stand alone mode can be use at any time. There are certian times when the decoder must operate in conjunction with the encoder. For example, when J.52 line bonding is used.

The realtime decoder status bits for the ISO/MPEG algorithm are displayed by the **DRS** command. The status bits displayed are the ISO frame header bits which are set by the encoder (See Encoder Header).

The audio output of the decoder can be muted by the **DMU** command.

The decoder audio output is controlled by the **DCS** command. This allows the swapping fo the left and right channel audio output. It also allows the left channel to be copied to the right channel (left channel mono) or the right channel to be copied to the left channel (right channel mono).

The decoder Digital to Analog (DA) converter can be calibrated by the **DDA** command. This calibration process insures that the DA converter is operating properly.

The ISO/MPEG scale factors can be protected by a CRC. This feature is controlled by the **DSP** command. In general, it is better to use scale factor protection if the data channel is noisy (high BER). If scale factor protection is enabled in the decoder, it must also be enabled in the encoder (**ESP**) or else the decoder output will mute.

The decoder can be instructed to decode only ISO/MPEG layer 2 bit streams by the **DCO** command. This is useful for determining if the incomming bitstream is fully ISO/MPEG compliant.

The decoder provides a method of generating test tones. The frequency and level of these tones are controlled by the **DMD** command.

If the decoder is operated in the stand alone mode (by setting **DIN** to YES), then there are several commands which must be set to determine the operation of the decoder. The first of these is the decoder bit rate. This is the compressed data rate and is set by the **DBR** command.

The **DLI** command is used to set the line format and is set by the **DLI** command. The **DLI** command instructs the decoder how to interperate the incomming compressed digital data. For example, if the incomming data is only present on digital interface 1 (DIF 1) then **DLI L1** instructs the decoder to receive the data on that line.

The decoder algorithm is another parameter which is meaningful only in the decoder independent mode. The **DAL** command sets the decoder algorithm. This forces the decoder to operate utilizing a particular decompression algorithm.

### 3.4 Digital Interface

Associated Remote Control Commands

CDT    Set state of the DTR/CON line

349

3-2

BAD ORIGINAL

**CIF Set digital interface type**

The compressed audio digital interface (DIF) type is defined by this command. There are several types of digital interfaces. The types are TA and non-TA types. A TA type of digital interface is one that is capable of connecting to the ISDN line and can dial. The states of a TA type interface are

- DISCONNECTED
- DIALING
- CONNECTED

For cdqPRIMA models with the LED display, the states of the digital interface is shown by the 6 DIF LED's. If the LED is dark, then the state of the DIF is disconnected. If it is blinking, then the DIF is dialing and if the LED is illuminated, then the DIF is connected.

A non-TA interface is always in the connected state and there are several types of these interfaces. A list of these interfaces is shown below.

- X.21
- RS422
- V.35

The RS422 and X.21 have the same voltage levels and thus are both on the same interface card. This distinction between them is made by setting jumpers on the card.

The V.35 standard specifies different voltage levels and hence must use different type of line interface IC's. The interface card used for this standard is different from the interface card for the RS422/X.21 standard.

The **CIF** command (and corresponding LCD command) is used to define the type of digital interface to be used.

On the non-TA interfaces, there is a signal designated DTR for the V.35 interface and CON for the X.21 interface. These are control lines from the cdqPRIMA interface card to the external terminal adaptor equipment. The levels of these lines are controlled by the **CDT** command. Some external ISDN and switch 56 TA's require that the DTR/CON line is asserted. The **CDT** command provides an easy method of controlling the DTR/CON line.

### 3.5 Encoder

#### Associated Remote Control Commands

- EAD Calibrate AD converter
- EAI Set encoder audio input source

350

BAD ORIGINAL

3-3

EAL	Set encoder algorithm
EAM	Set encoder algorithm mode
EBR	Set encoder bit rate
ELI	Set encoder digital lines format
ESP	Set scale factor protection
ESR	Set encoder sampling rate

The compressed digital audio encoder is controlled by the above commands. If the decoder is dependent on the encoder (**DIN** NO), then some of these encoder commands also control the decoder.

The source of the audio input is controlled by the **EAI** command. The source may be the analog inputs or the digital AES/EBU inputs. The analog input AD converter is calibrated by the **EAD** command. This calibration is done at power-up but can be done at any time. The calibration process removes the effect of any DC voltage offset present at the input of the AD converter. This has a minor positive effect on the audio compression algorithm.

The encoder audio compression algorithm is set by the **EAL** command. If the algorithm is one of the ISO/MPEG types, then the **EAM** command set the mode to mono, dual mono, joint stereo or stereo. The digital audio sampling rate is controlled by the **ESR** command while the compressed audio bit rate is controlled by the **EBR** command.

The **ELI** command is used to control how the compressed digital audio bit stream is transmitted. For example, if **ELI** L1 is used, then the compressed output bits are sent out digital interface (DIF) 1. Scale factor protection (**ESP**) is used for ISO/MPEG types of bitstreams. Scale factors are the levels of the digital audio signal within a sub-band. There are 32 sub-bands and the scalefactors change the level over a 120 dB range. An error on any scale factor will cause a perceptible impairment in the audio. To prevent this, scalefactor protection can be inserted at the encoder and if the decoder is capable of recognizing it, then the decoder can perform a concealment operation to repair the damaged scalefactor. If the decoder does not know about scale factor protection, the audio is decoded and any damaged scalefactors cause an impairment. If **ESP** has enabled scalefactor protection, the far end decoder must enable scale factor correction by the **DSP** command.

### 3.6 Encoder Header

#### Associated Remote Control Commands

ECR	Set encoder copyright bit in header
EEP	Set encoder emphasis bit in header
EOR	Set encoder original bit in header
EPR	Set encoder protection bit in header

When utilizing the CCSO, CCSN or MPEG audio compression algorithm, there are certain flags which may be set in the header. These bits can be used by the decoder. These bits are defined below and the command used to set the bit is shown in parenthesis.

- Copyright (**ECR**)
- Emphasis (**EEP**)
- Original (**EOR**)
- Protection (**EPR**)

The cdqPRIMA decoder reads these bits and displays them. The state of these status bits can be seen by executing the **DRS** or the **CST** command.

351

3-4

BAD ORIGINAL

ON

### 3.7 Front Remote Control

#### Associated Remote Control Commands

CFB	Set front panel remote control baud rate
CFP	Set front panel remote control protocol usage
CFE	Set front panel remote control command response echo

Front panel remote control is provided on all models except the 110 and 210. Front panel remote control allows computer access to all the internal functions of the cdqPRIMA. Front panel remote control is especially useful for applications which need quick access to the cdqPRIMA via a palm top computer. This frequently occurs in control rooms in which there are many cdqPRIMA's in equipment racks.

The baud rate of the front panel access is set by the **CFB** command.

The protocol for this interface is defined by the **CFP** command. There are two possible protocols for communication with the cdqPRIMA. This first is simple ASCII messages which can be generated by any terminal emulator communications package. The second method of communications is via protocol protected messages. In this case, the simple ASCII message is surrounded by a header at the beginning of the message to specify the byte length of the message and other parameters and a CRC is appended to the end of the message for error control. The details of the protocol is covered in the chapter entitled cdqPRIMA Remote Control Protocol.

When downloading the cdqPRIMA, it is possible to turn off the command echo. This speeds up the download process at the expense of seeing the command echo. The command echo can be turned off by utilizing the **CFE** command.

### 3.8 Headphones

#### Associated Remote Control Commands

CHV	Set headphone volume level of current device
DHV	Set decoder headphone volume level
EHV	Set encoder headphone volume level
CHP	Set headphone audio source

The front panel headphone output can be connected to either the encoder input signal (after the A/D converter) or to the decoder output (before the D/A converter) by the **CHP** command. The headphone can listen to the stereo signal (left channel to left earphone and right channel to right earphone) or the left channel only (left channel to left and right earphone) or the right channel only (right channel to left and right earphone).

The headphone volume may be adjusted by the **CHV**, **DHV** and **EHV** commands. The volume of the encoder and decoder are adjusted separately. There are not separate adjustments for the left and right channels. The volume level is from 0 to 127 arbitrary units with 0 being mute and 127 being the loudest.

352

3-5

BAD ORIGINAL



### 3.9 Help

#### Associated Remote Control Commands

CQQ	Print command summary for common commands
DQQ	Print command summary for decoder commands
EQQ	Print command summary for encoder commands
HELP	Print all help commands
MQQ	Print command summary for maintenance commands

There are 4 categories of commands. These are

- Common commands
- Decoder commands
- Encoder commands
- Maintenance commands

Executing CQQ, DQQ, EQQ or MQQ lists a command summary for each of the command groups.

The commands are arranged in functional groups and these groups are displayed by executing the **HELP ?** command. A summary of each command group is shown by executing **HELP xxx** where **xxx** is a number between 1 and 30.

Each command has its own help. This help is displayed by typing **HELP cmd** or **cmd HELP** where **cmd** is any three character command.

### 3.10 Hot Keys

#### Associated Remote Control Command

CHK	Define hot key
-----	----------------

On certain models (the 230), user definable hot keys are available. These keys allow the user to attach a cdqPRIMA remote control command to a key. Once the command has been attached to the key, a depression of the key causes the command to execute. See the **CHK** command for a detailed explanation of the syntax of this command.

### 3.11 Loop Back

#### Associated Remote Control Commands

CBR	Set loopback bit rate
CLB	Set loopback on a digital data interface
CSL	Set system loopback



The **cdqPRIMA** has two types of loopback. The first type is a system loopback and the second is a digital interface loopback. The system loopback is an internal loopback and is set by the **CSL** command. It loops all the digital interfaces internally with one command.

The **CLB** command is used to set the loopback on each digital interface module. Some modules such as the X.21 and the V.35 card respond to this loopback. The TA cards generally do not respond to the **CLB** command.

When the **CSL** command is set to loopback (**LB**) then the internal clock is used as to supply the digital data clocks. The clock rate of this clock is set by the **CBR** command. The bitrate set by this command only applies when **CSL** is set th **LB**. When **CSL** is set to **LB**, the **EBR** and the **DBR** commands are ignored.

### 3.12 Maintenance

#### Associated Remote Control Commands

<b>CDF</b>	Set default parameters
<b>MCP</b>	Set connect port
<b>MSY</b>	Synchronize RAM and BBM
<b>MVN</b>	Print software version numbers
<b>MWP</b>	Set watch port

All of the **cdqPRIMA** parameters can be set to the factory default state by executing the **CDF** command. The psychoacoustic parameters and the speed dial numbers are not reset by the **CDF** command. The **CDF** command is also executed at power up when the 0 key on the front panel is depressed until the **TOTAL RESET OF ALL PARAMETERS** is displayed.

The **MCP** is used to connect the remote control port to an internal uart and monitor traffic to and from the specified serial port. It is used for debugging only and should be used only with the guidance of experienced technical support personal.

When commands are executed, the command argument is written to non-volatile RAM. For example if the **ELI L1** command is issued, then the **L1** is remembered in non-volatile RAM and if power is removed, the setting is remembered. When power is restored, the **ELI L1** command is read from non-volatile RAM and executed in an attempt to restore the **cdqPRIMA** to the state that existed before power was removed. Some commands write their argument to a cache which is later written to non-volatile RAM. The execution of the **MSY** command causes all entries to be written to non-volatile RAM immediately. This should be done just before powering down to insure that all parameters are in non-volatile memory.

The **MVN** command can be used to print the version number of the various software modules. It also prints the module checksum and length.

The **MWP** command is used for software debugging only. It should be used under the direction of an experienced maintenance technician.

354

3-7





### 3.13 Out Of Frame Detector

#### Associated Remote control Commands

MOC	Display OOF counter
MOD	Set OOF down count rate
MOL	Set OOF count rate limit
MOR	Reset OOF counter
MOU	Set OOF up count rate

The out of frame detector provides a method of monitoring the number of framing errors that occurred in the digital transmission path. The out of frame rate detector is used when ISO/MPEG type of frames are enabled by the **EAL** command. When each ISO/MPEG frame is received (every 24 milliseconds for 48 kHz sampling), the header CRC is checked for validity. If the frame header has valid framing bits, then the OOF counter is incremented by a OOF up count (any number from 0 to 9). If the frame header bits are invalid, then the OOF counter is decremented by the OOF down count (any number from 0 to 9).

If the OOF up count is set to 1 (by the **MOU** command) and the down count is set to 0 (by the **MOD** command), then the OOF counter counts the total number of frames in error. If the up counter is set to 2 and the down count is set to 1, then the OOF counter is sensitive to burst errors but not random errors.

The OOF counter is compared to the OOF threshold (set by the **MOL** command) to see if the counter is above or below the threshold. Actions such as closing a relay, dialing a phonenummer or lighting a LED or displaying a scrolling message can be taken.

The OOF counter can be reset to 0 by the **MOR** command.

The current contents of the OOF counter can be displayed by the **MOC** command.

### 3.14 Peak Detector

#### Related Remote Control Command

MPD	Display peak detector level
-----	-----------------------------

The **MPD** command is used to display the highest peak level for the encoder or the decoder, right or left channel. After executing this command, the highest peak level is set to -150 dBu and is updated by the the audio input. The peak level is retained even after all audio has stopped and can be read once by executing the **MPD** command.

### 3.15 cdqPRIMA Logic Language

#### Associated Remote Control Commands

CAR	Clear the latched value of the action word
CCT	Cancel timer

355

3-8



CEA	Set event to action logic
CEV	Print event inputs
CLA	Print latched value of the action word
CRA	Print realtime value of the action word
CTM	Set timer timeout duration
CVA	Define virtual action
ELU	Set link message update rate
ESW	Set a simulated switch

The cdqPRIMA has a rich language for mapping input events such as high BER into actions such as relay contact closure.

A detailed description of the cdqPRIMA Logic Language (PLL) is given in the cdqPRIMA Logic Language section.

The inputs to the Event to Action interpreter are displayed by the **CEV** command. These input events may be physical inputs such as input optical isolators or logical input such as computer generated switch closures (see the **ESW** command).

The mapping of input events into output actions is controlled by the **CEA** command. This command is described fully in the cdqPRIMA Logic Language section.

The real-time value of the Action Word is displayed by the **CRA** command while the latched value of the Action Word is displayed by the **CLA** command. The latched Action word values are reset via the **CAR** command.

Action Words are the output of the Event to Action logic which is controlled by the cdqPRIMA Logic Language (PLL). See the section entitled cdqPRIMA Logic Language for further details of the PLL. Actions are real and virtual. The real action are thing such as lighting a relay or virtual actions such as executing a remote control command (see the **CVA** command).

The other virtual action is the starting of a timer (see the **CTM** command). The expiration of a timer is an input event. Timers can be cancelled by the **CCT** command.

Actions can be exported to a far end cdqPRIMA. This exported action appears as an input event to the far end cdqPRIMA. The exported actions are transmitted to the far end at a rate governed by the **ELU** command. The actions are exported repeatedly in an attempt to insure their arrival at the far end even in the presence of a noisy digital communications channel.

### 3.16 Quiet Detector

#### Associated Remote Control Commands

MQC	Display quiet detector level time left
MQD	Display quiet detector level
MQL	Set quiet detector level
MQT	Set quiet time duration

356

3-9

BAD ORIGINAL

There are 6 silence detectors (quiet detectors). These are

- encoder left channel input
- encoder right channel input
- encoder stereo input
- decoder left channel output
- decoder right channel output
- decoder stereo output

A stereo silence detector uses the greater of the left or right channel signal in the silence determination.

At .1 second intervals the audio levels of the encoder and decoder, left and right channels are measured. If the level is below the value set by the **MQL** command for a period of time set by the **MQT** command, then the channel is said to be silent. When a channel is silent, the silent event input is set to true. The value of the silence event may be used as input the the Event to Action logic interpreter.

The current value of any of the 6 quiet detectors can be displayed by the **MQD** command.

The time left before silence is detected can be displayed by the **MQC** command.

### 3.17 Psychoacoustic Parameter Adjustment

Associated Remote Control Commands

EPD	Get default psychoacoustic parameter table number
EPL	Load psychoacoustic parameters from flash
EPP	Set psychoacoustic parameter
EPS	Store psychoacoustic parameters in flash
EPT	Assign psychoacoustic parameter table
EPY	Set psychoacoustic parameter type

There are 32 psychoacoustic parameters which control the cdqPRIMA. The manipulation of these parameters is discussed in the section Psychoacoustic Parameter Adjustment.

### 3.18 Remote Control

Associated Remote Control Commands

CID	Set RS485 remote control ID
CPC	Set remote control protocol usage
CRB	Set remote control baud rate
CRI	Set remote control type
CRE	Set remote control command response echo

357



Rear panel remote control is provided on all models. Rear panel remote control allows computer access to all the internal functions of the cdqPRIMA. Rear panel remote control is especially useful for applications which need permanent access to the cdqPRIMA via a control computer. This frequently occurs when the cdqPRIMA is remotely located from the control room.

The rear panel remote control electrical interface may be either RS232 or RS485. The RS485 interface may be either the 2 or 4 wire interface. The choice of the electrical interface is controlled by the **CRI** command.

If protocol protected messages are used to control the cdqPRIMA, then the message must have a destination id. This id is set by the **CID** command.

The baud rate of the rear panel remote control port is set by the **CRB** command.

The protocol for this interface is defined by the **CPC** command. There are two possible protocols for communication with the cdqPRIMA. This first is simple ASCII messages which can be generated by any terminal emulator communications package. The second method of communications is via protocol protected messages. In this case, the simple ASCII message is surrounded by a header at the beginning of the message to specify the byte length of the message and other parameters and a CRC is appended to the end of the message for error control. The details of the protocol is covered in the chapter entitled cdqPRIMA Remote Control Protocol.

When downloading the cdqPRIMA, it is possible to turn off the command echo. This speeds up the download process at the expense of seeing the command echo. The command echo can be turned off by utilizing the **CRE** command.

### 3.19 Security

Associated Remote Control Command

CPW Set user's security status

CPW Set user's security status

### 3.20 Software Maintenance

Associated Remote Control Command

CVN Print software version number

The software version number of any flash object can be displayed via the **CVN** command. Each internal algorithm is called a Flash Object. Each Flash Object has its own internal version number.

358

3-11

BAD ORIGINAL

### 3.21 Speed Dialing

#### Associated Remote Control Commands

CDS	Delete a speed dial number
CSC	Clear all speed dial entries
CSD	Speed dial a number
CSE	Enter a number in speed dial directory
CSF	Display first of speed dial entry
CSN	Display next of speed dial entry

The speed dial feature allows the entry of 256 system configurations consists of up to 6 telephone number, the sampling rate, line format and a description. Speed dial entries are entered by either the **CSE** command or the **SDSET** button. Each of the speed dial entries is given a speed dial id (3 digit number). A speed dial entry is activated by either the execution of the **CSD** command or the front panel **SDIAL** button.

All speed dial entries can be deleted via the **CSC** command while a single speed dial entry can be deleted by the **CDS** command.

The entire speed dial list may be displayed by first typing **CSF** to display the first speed dial entry and then entering **CSN** repeatedly to display the subsequent speed dial entries. The speed dial entries are displayed in alphabetical order by description.

### 3.22 Status and Level Display

#### Associated Remote Control Commands

CLI	Set LED display intensity
CLM	Display LED message
CVU	Set level meter mode

A front panel LED display is provided on all models except the 110 and the 210. This front panel display can be used for various functions. The **CVU** command is used to set the measurement mode. The normal mode is the level indication mode in which the average and peak input signal is displayed. The stereo image can be displayed as well as the left/right channel correlation.

The level mode of operation is the usual level indicator mode. The right hand side of the level display is labeled 0 dB and each LED to the left represents 2 dB weaker signal. The right hand 5 LED's are red, the next 5 LED's to the left are yellow and the last 10 LED's on the left are green. Thus the 20 LED's represents a 40 db range.

The far right LED has a reversed arrow display to indicate that the input or output is at the maximum level of 0 dB. The VU meter is labeled with 0 at the maximum because the input amplifiers may be different values. For example, the standard input amplifier on the cdqPRIMA allows a maximum input of +18 dBu. If a sinewave with a peak to peak level of +18 dBu is input to this amplifier module, the peak level LED will read 0. A 0

359

3-12

BAD ORIGINAL

level of the level LED means that the input is at the maximum allowed value. The output LED display is similar.

The level display consists of an encoder and a decoder section, each with a left and a right channel display. Each channel display consists of a single LED representing the peak value and a solid group of LEDs representing the average value of the input audio.

If the stereo image display is selected, the scale below the display must be used. This scale shows the relative location of the stereo image. If the image is centered, then the single LED is illuminated above the C. If the image is to the right then the LED is displayed toward the L. This display is useful when the gains of the left and right channels must be balanced for stereo signals.

The stereo correlation display is indicated by a double LED illumination. The correlation display is useful to detect if the input signal can be mixed to mono. A correlation from 0 to +1 indicates that there is mono compatibility while a stereo correlation near -1 indicates that the left and right signals are out of phase and cannot be mixed to mono.

The CLM command allows a scrolling message to be displayed on the front panel LED display. This is useful to alert a remote location of an upcoming feed or provide a cue.

The CLI command is used to set the intensity of the LED display. The display is broken into 3 groups and the intensity of each group can be controlled. This allows instant focus on one group by dimming the intensity on the other groups.

### 3.23 Status

Associated Remote Control Command

CST Report CODEC status

A general system status is provided when the CST command is executed. This status is intended to be a snapshot of all system functions.

### 3.24 System Setup

Associated Remote Control Command

CDF Set default parameters

The CDF command is used to restore the factory defaults for everything except the psychoacoustic parameters and the speed dial numbers. To test the unit that seems to be confused, one can issue the CDF followed by the CSL LB commands to set the defaults and set the system into loopback. See the CDF command for a list of the system defaults.

360

3-13



### 3.25 Terminal Adaptor

#### Associated Remote Control Commands

CAA	Set TA auto answer mode
CAC	Set TA auto-reconnection state
CAD	Auto dial phone numbers
CCR	Clear TA digital interface connect time
CCS	Print TA digital interface connect time
CDC	Real-time display TA digital interface connect time on LCD
CDI	Dial TA phone number
CHU	Hang up a line or lines
CLD	Set ID for a Terminal Adaptor
CSI	Set SPID for a Terminal Adaptor
CSW	Set switch type
CTC	Connect to a TA control port
CTE	Set TA remote control command response echo
CTP	Set TA remote control protocol usage
CTO	Set TA dialing timeout

The ISDN type of digital interface module allows access to the ISDN network. There are several types of ISDN TA's available for the cdqPRIMA.

The TA101 provides 1 BRI (2 \* 64 kbs) access to the network. This TA requires different ROMS for different countries. The TA201 and TA202 ROMS have onboard FLASH memory with the switch configurations for different countries. Contact the factory to obtain the proper ROM for your country if you are utilizing a TA101 TA.

If the TA is operated in North America, the the switch type (**CSW**), line ID (**CLD**) and line SPID (**CSI**) must be entered before any calls can be placed. See Appendix A for TA101 setup information.

A direction connection with the TA is performed by the **CTC** command. This mode of operation is useful because it allows the lowest level of control over the TA. When the **CTC** command is used, then all of the low level TA commands are available. Consult the factory for a description of these low level commands.

The **CAA** command can be used to set the TA into the auto answer mode. If the TA is not in the auto-answer mode, then it will not accept any incoming calls.

An individual line may be connected by utilizing the **CDI** command. This command allows dialing individual ISDN lines at either 56 or 64 kbs. Once a call has been placed to the far end, a timeout is in effect waiting for the far end to answer. This timeout is set by the **CTO** command.

Once a call has been placed, by the **CSD** or **CDI** command, the line or lines may be "hung up" by the **CHU** command. This command disconnects a connected line.

If a connection is made to a far end TA and the connection is lost, it is possible to have the cdqPRIMA automatically re-establish the connection. This is done by the **CAC** command.

The cdqPRIMA allows the display of the time the line has been connected of any one of the 6 digital interfaces. This is useful for estimating the cost of the connection. The **CDC** command is used to display the connect time on the LCD screen. The current time connected for any of the 6 lines can be printed on a remote control terminal by the **CCS** command. The connect time counter can be set to zero at any time by the **CCR** command.

The cdqPRIMA allows direct connection over ISDN into the ISDN remote control port. This allows complete remote control including software down load from a far end cdqPRIMA via ISDN. The **CTP** command is used to enable or disable command protocol usage over the ISDN line while the **CTE** command is used to control the command response echo.

### 3.26 Test

#### Associated Remote Control Commands

<b>MTM</b>	Perform a test measurement
<b>MET</b>	Enable hardware tests

<b>MET</b>	Enable hardware tests
------------	-----------------------

The cdqPRIMA can be used to perform various tests on external equipment. These tests are controlled by the **MTM** command.

### 3.27 Time Code

#### Associated Remote Control Commands

<b>CTI</b>	Set Time Code readout source
<b>CTL</b>	Print last Time Code received
<b>CTS</b>	Print Time Code speed
<b>CTT</b>	Enable/disable Time Code

**SMPTE** time code is an optional feature of the cdqPRIMA. **SMPTE** time code is read by the optional reader, converted into a digital bit stream, muxed with other data and send to the decoder as ancillary data. The mux mode of ancillary data (**CAN 2**) must be used and the audio algorithm cannot be **G.722** in order to use **SMPTE** timecode.

The **SMPTE** timecode reader and generator in the cdqPRIMA automatically sense the the input timecode rate with no external control necessary. The cdqPRIMA allows the user to transmit timecode simultaneously with the audio and thus the cdqPRIMA is the perfect unit for studios utilizing audio/video timecode.

362

3-15

BAD ORIGINAL





SMPTE timecode utilizes approximately 2.4 kbs of digital bandwidth. This small overhead allows transmission of timecode even at bits rates of 56 and 64 kbs. If the timecode input is removed, then no digital bandwidth is used. It may be inconvenient to remove the timecode input and the **CTT** command can be used to enable/disable the transmission of timecode. Turning timecode off with the **CTT** command has the same effect as removing the timecode connector from the rear of the cdqPRIMA.

The **CTS** command is used to print the timecode speed.

The current time code may be displayed by the **CTI** command. The displayed timecode may be the timecode input to the encoder or the timecode received by the decoder.

The last timecode received may be displayed by the **CTL** command.

### 3.28 Timing

Associated Remote Control Commands

DES	Decoder AES timing
ETI	Encoder timing
DDO	Set digital output sampling rate
DTI	Decoder timing

The timing of the encoder and decoder can be controlled by various commands. These commands are documented in the Digital Timing Section of this manual.

### 3.29 Misc

Associated Remote Control Command

COM	Comment command
-----	-----------------

The **COM** command performs nothing and is useful for inserting comments in command scripts.

### 3.30 Download/Boot

Associated Remote Control Commands

MBM	Boot the cdqPRIMA from ROM
-----	----------------------------

Normally the cdqPRIMA executes its software from the FLASH memory. If this FLASH memory needs to be updated, then it must operate out of the boot ROM.

The **MBM** command is used to force the cdqPRIMA to operate from the ROM boot. This is required when downloading new software into the cdqPRIMA.



### 3.31 Sync Ancillary Data

#### Associated Remote Control Commands

DSB	Set decoder synchronous ancillary data rate
DSC	Set decoder synchronous ancillary data clock edge
ESB	Set encoder synchronous ancillary data rate
ESC	Set encoder synchronous ancillary data clock edge

The synchronous ancillary data commands allow the bit rate (**DSB** and **ESB**) to be set for the decoder and encoder. The clock edge (low to high or high to low) for clocking valid data can also be set for the encoder and the decoder (**DSC** and **ESC**).

364

3-17

BAD ORIGINAL

## 4. Operation

### 4.1 Quick Start

The cdqPRIMA is shipped from the factory configured for loopback operation. This means that at power up, the cdqPRIMA should operate correctly and pass audio from the input to the output. The settings for the encoder are given under the **CDF** command but they are summarized below.

<i>parameter</i>	<i>value</i>
bitrate	128 kbs
algorithm	MPEGL2
mode	joint stereo
sampling rate	128
encoder line format	L1
decoder set to independent	NO

Table 4-1

Summary of default setups

### 4.2 Front Panel Displays

#### 4.2.1 Character Display (Models 110, 120, 210 & 220)

The LCD display for the cdqPRIMA models 110, 120, 210 and 220 models is a 2 line by 16 characters. This display is used for all responses to front panel user commands as well as spontaneous messages such as incoming call connect messages.

#### 4.2.2 Graphics Display (Model 230)

The cdqPRIMA model 230 has a graphics display which allows 8 rows of 40 characters or 240 by 64 pixels. When operating in the character mode, the display functions in a manner similar to the cdqPRIMA 110. The graphics mode is used for graphical display of measurement information.

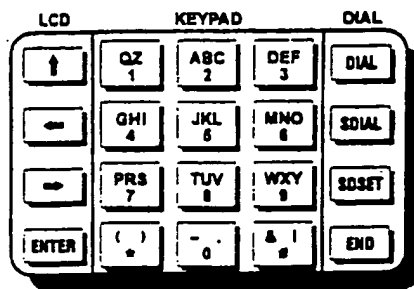
### 4.3 Front Panel Controls

365

4-15

BAD ORIGINAL





**Figure 1**  
Model 110 & 210 keypad

#### 4.3.1 Cursor Keys (All Models)

The 4 keys under the LCD label are used to control the cursor. They are

- UP ARROW
- LEFT ARROW
- RIGHT ARROW
- ENTER

The up arrow key is used to move up the menu tree. This key is also use on power up to force entry into the ROM boot mode which is used for local downloading software. The up arrow key is also used to terminate any graphical measurements which are in progress.

The left and right arrow keys are used to move to the right and left in the menu tree.

The ENTER key is used to execute the menu tree entry enclosed within the square brackets ( [ ] ).

#### 4.3.2 Dial Keypad (All Models)

The dial keypad consists of the 12 keys under the KEYPAD label. These keys forma a general purpose alpha-numeric keypad. Different commands enable different characters on these keys. For example, dialing commands only enable the numeric selections for these keys. When cdqPRIMA Logic Language commands are entered, all of the keys are enabled. By depressing the 2 key repeatedly, the A, B and C keys are displayed. In such multi-character modes are enabled, the right and left arrow keys are used to move to the right and left on the current line. The Enter key is used to accept the entire entry.

366



#### 4.3.3 Dial Setup Keys (All Models)

The 4 keys below the **DIAL** label are used for dialing. They are

- **DIAL**
- **SDIAL**
- **SDSET**
- **END**

The dial key allows the dialing of a single ISDN line. Before dialing can be attempted, the Digital InterFace (DIF) must be defined by utilizing the CIF command. The DIF must contain a TA type of Digital Interface Module (DIM) such as a TA101.

Depressing the **DIAL** key begins the dialing sequence and the LCD display will prompt the user for the bit rate and telephone number. Once the enter key is depressed denoting the entry of the phone number, then the dialing operation begins and the DIF LED begins to blink indicating that the phone is dialing. When the light becomes solidly on, the connection is established. The calling status is also displayed on the LCD screen.

The **SDIAL** key is used to speed dial a destination. After depressing **SDIAL**, the LCD screen prompts for the 3 digit speed dial number which is terminated by depressing the **ENTER** key. The parameter required by this operation is described in the CSD remote control command.

The **SDSET** key is used to setup a speed dial entry. Depressing this key produces a series of prompts on the LCD display to enter the speed dial parameters. The parameters to be entered are described in the CSE remote control command.

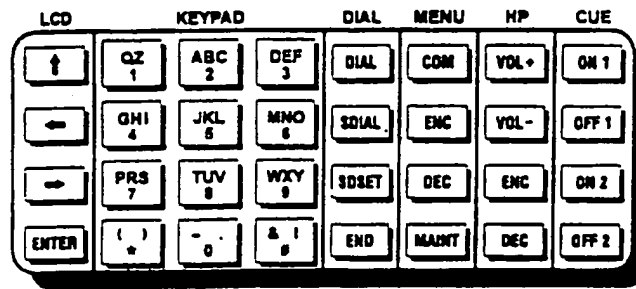
The **END** key is used to terminate a connections made by the **DIAL** and **SDIAL** keys. Depressing this key allows all lines or a single line to be dropped. See the CHU command.

367

4-20

BAD ORIGINAL





**Figure 2**  
Model 120 & 220 keypad

#### 4.3.4 Menu Keys (Models 120, 220 & 230)

The 4 keys under the MENU label are used to quickly move the one of the 4 main branches of the menu tree. These branches are

- COM      Commands common to the entire unit
- ENC      Commands for the encoder
- DEC      Commands for the decoder
- MAIN     Maintenance commands

#### 4.3.5 Headphone Keys (Models 120, 220 & 230)

The 4 keys under the HP label are used to control the output of the front panel headphone jack. These keys are

- VOL+
- VOL-
- ENC
- DEC

The keys labeled ENC and DEC are used to select the encoder and decoder respectively. If the ENC button is depressed, the input signal to the encoder section is output to the headphone. If the DEC button is depressed, the decoder output is present at the headphone jack. There are 4 LED's under the label HP STATUS which are controlled by the ENC and DEC push buttons. If the ENC button is depressed, one or both of the encoder headphone LED's illuminate. When the ENC button is first depressed, the output of the left and right channels are output to the left and right earphones. If the ENC

368

4-21

BAD ORIGINAL

button is depressed again, the encoder left channel LED is illuminated and the input to the encoder left channel is output to both the left and right channel headphones. If the ENC button is depressed again, the left channel HP LED is illuminated and the signal which is input to the right channel of the encoder is connected to both the left and the right channel of the headphones. A similar action occurs when the DEC button is repeatedly depressed.

The VOL+ and VOL- buttons control the volume of the headphone output. Depressing the VOL+ increases the headphone volume while depressing the VOL- button decreases the headphone volume. The headphone volume level ranges from 0 (mute) to 127 in arbitrary volume units (approximately 1 dB steps).

The volume buttons control the left and right channels simultaneously but the encoder and decoder output signals have separate volume levels which are active when the ENC and the DEC buttons are depressed.

If the headphone volume is set too high, distortion may occur.

#### 4.3.6 Cue Keys (Models 120, 220 & 230)

The 4 buttons under the CUE label are general purpose front panel switches. These 4 buttons represent 2 switches. These two switches can be either on or off. For example, depressing the ON1 button, causes switch 1 to turn on while depressing the OFF1 button causes switch 1 to turn off. The corresponding action occurs for the ON2 and OFF2 buttons for switch 2. Switch 1 and switch 2 are connected to CI1 and CI2 (see the section on cdqPRIMA Logic Language).

The default setup of the cdqPRIMA assigns switch 1 (ON1 and OFF1 buttons) to a sending a cue from the near end to a far end. Depressing the ON1 button, causes the SCUE1 LED to illuminate indicating that cue 1 is being sent. The far end cdqPRIMA will illuminate its RCUE1 LED indicating that it received this cue. Depressing the OFF1 button causes the SCUE1 LED to extinguish indicating that there is no cue 1 being sent.

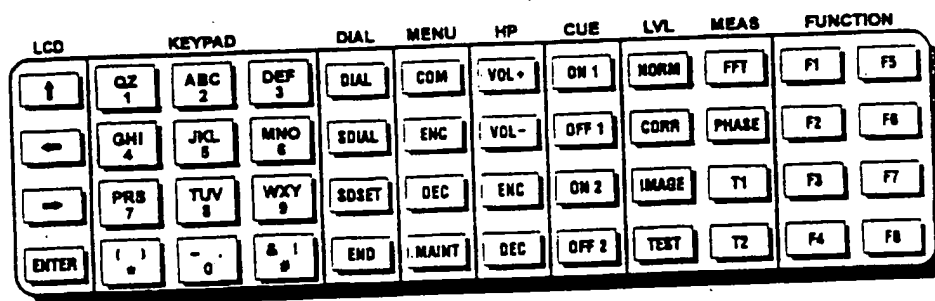


Figure 3  
Model 230 keypad

369

4-22



#### 4.3.7 Level Control Keys (Model 230)

The 4 keys below the LVL label on the front panel are used to control the audio level LED display. The keys are labeled

- NORM
- CORR
- IMAG
- TEST

Depressing the **NORM** key causes the audio level LED's to display the average and peak levels. Each LED represents 2 dB and the signal corresponding to the maximum input is labeled 0 dB.

Depressing the **CORR** key causes the level LED's to display the stereo correlation. The values for the left/right correlation are +1 to -1 where +1 indicates the left and right channels are exactly in phase. A correlation of -1 indicates that the left and right channels are exactly out of phase. In phase stereo signals may be mixed into a mono signal.

Depressing the **IMAG** key causes the level LED's to display the stereo image of the left and right channel. If the power of the left and right channels are the same, then the stereo image will be in the center above the stereo image label C. If the power of the right channel is more than the left channel, the stereo image LED will move to the right indicating the stereo image has moved to the right.

Depressing the **TEST** button causes all the LED's to illuminate for a few seconds to allow visual inspection of all the LED's.

#### 4.3.8 Measurement Keys (Model 230)

The 4 measurement keys

- FFT
- PHASE
- T1
- T2

are use for graphics measurements. The results of all these measurements are displayed on the graphics display.

370

BAD ORIGINAL

4-23



The key labeled **FFT** is used to enable the real-time spectrum analyzer of the signal which is input to the left channel of the encoder.

The **PHASE** key is used to display a real-time phase display of the left and right channels.

The **T1** and **T2** keys are currently not assigned to any measurement function.

#### 4.3.9 Function Keys (Model 230)

The 8 keys labeled **F1** through **F8** are user definable function (hot) keys. Any remote control command may be attached to any of these keys. See the **CHK** command for instructions on how to define one of these hot keys.

#### 4.4 Front Panel Indicators

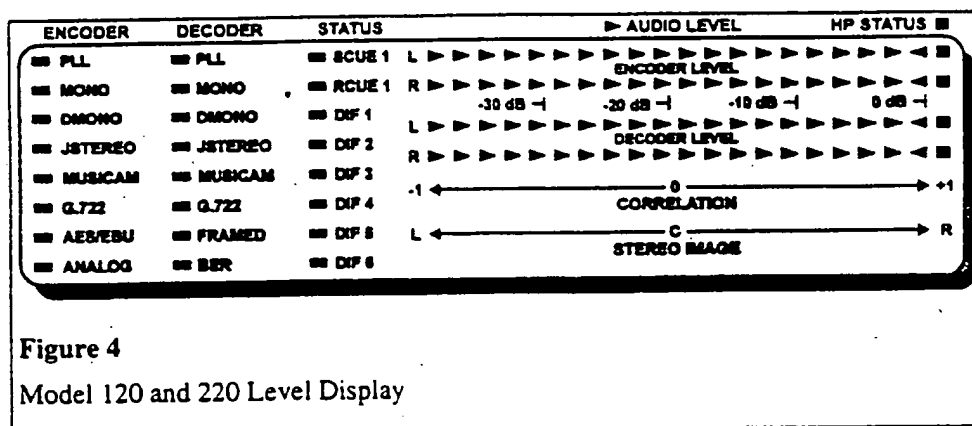


Figure 4

Model 120 and 220 Level Display

#### 4.4.1 Model 120 and 220

##### 4.4.1.1 Encoder

###### 4.4.1.1.1 PLL

This LED illuminates green when the encoder phase locked loop is locked. This LED must be on for proper operation.

###### 4.4.1.1.2 MONO

This LED illuminates yellow when the ISO/MPEG frame is transmitting a mono signal. This led is also illuminated when G.722 is being transmitted.



**4.4.1.1.3 DMONO**

This LED illuminates yellow when the ISO/MPEG frame is transmitting dual mono.

**4.4.1.1.4 JSTEREO**

This LED illuminates yellow when the ISO/MPEG frame is transmitting in the joint stereo mode. If the MONO, DMONO and JSTEREO LED's are all extinguished, then the encoder is outputting stereo frames.

**4.4.1.1.5 MUSICAM**

This LED illuminates yellow when CCS MUSICAM or ISO/MPEG frames are being transmitted.

**4.4.1.1.6 G.722**

This LED illuminates yellow when G.722 audio compression is being transmitted.

**4.4.1.1.7 AES/EBU**

This LED illuminates yellow when the input audio source is from the rear panel AES/EBU, SPDIF or optical inputs.

**4.4.1.1.8 ANALOG**

This led illuminates yellow when the input audio source is from the rear panel analog XLR connectors.

**4.4.1.2 Decoder****4.4.1.2.1 PLL**

This LED illuminates green when the encoder phase locked loop is locked. This LED must be on for proper operation.

**4.4.1.2.2 MONO**

This LED illuminates yellow when an ISO/MPEG frame is received and it is a mono signal. This LED is also illuminated when G.722 is being received.

**4.4.1.2.3 DMONO**

This LED illuminates yellow when an ISO/MPEG frame is received and its format is dual mono.

**372**

4-25

BAD ORIGINAL



#### 4.4.1.2.4 JSTEREO

This LED illuminates yellow when an ISO/MPEG frame is received and the type of frame is joint stereo. If the MONO, DMONO and JSTEREO LED's are all extinguished, then the decoder is receiving stereo frames.

#### 4.4.1.2.5 MUSICAM

This LED illuminates yellow when CCS MUSICAM or ISO/MPEG frames are received.

#### 4.4.1.2.6 G.722

This LED illuminates yellow when G.722 audio compression is received.

#### 4.4.1.2.7 FRAMED

This LED is used to indicate that the cdqPRIMA is receiving a properly framed signal. It illuminates green when the cdqPRIMA is framed.

#### 4.4.1.2.8 BER

This LED is used to indicate that a bit error has been detected. This LED illuminates red when a bit error has been received.

#### 4.4.1.3 Status

##### 4.4.1.3.1 SCUE1

Normally this LED illuminates when the ON1 button is depressed and is extinguished when the OFF1 button is depressed. Its normal meaning is that a cue has been sent to a far end decoder to be displayed on the far end RCUE1 LED.

The SCUE1 LED can be programmed to mean other things. See the chapter entitled cdqPRIMA Logic Language. for programming instructions.

##### 4.4.1.3.2 RCUE1

Normally this LED illuminates when cue 1 has been received from the far end encoder.

This LED can be reprogrammed to mean other things. See the chapter entitled cdqPRIMA Logic Language. for programming instructions.

##### 4.4.1.3.3 DIF1, DIF2, DIF3, DIF4, DIF5 and DIF6

There are 6 LED indicators for the digital interface status. These LED's can be in 3 states. These are

373

4-26

BAD ORIGINAL



- OFF (disconnected)
- BLINKING (TA dialing)
- ON (connected)

These states corresponding the interface status in parenthesis.

#### 4.4.2 Model 230

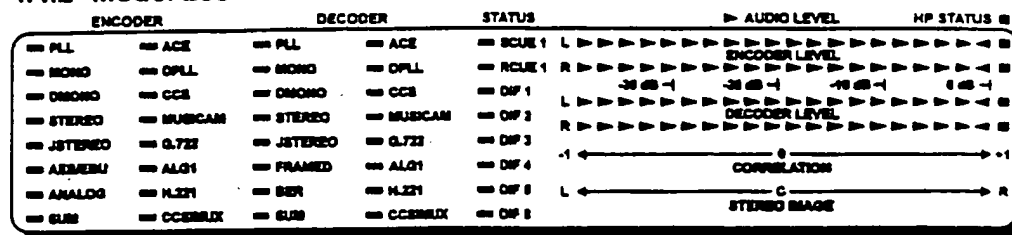


Figure 5

Model 230 Level Display

##### 4.4.2.1 Encoder

###### 4.4.2.1.1 PLL

See Model 120 display for a description.

###### 4.4.2.1.2 MONO

See Model 120 display for a description.

###### 4.4.2.1.3 DMONO

See Model 120 display for a description.

###### 4.4.2.1.4 STEREO

This LED illuminates yellow when an ISO/MPEG type of frame is sent and the mode of the signal is stereo.

###### 4.4.2.1.5 JSTEREO

This LED illuminates yellow when an ISO/MPEG type of frame is sent and the mode of the signal is joint stereo.

#### 4.4.2.1.6 AES/EBU

See Model 120 display for a description.

#### 4.4.2.1.7 ANALOG

See Model 120 display for a description.

#### 4.4.2.1.8 SUM

This LED illuminates when the encoder has detected an error. This LED is programmable and thus its meaning depends on the current definition. See the chapter entitled cdqPRIMA Logic Language.

#### 4.4.2.1.9 ACE

This LED illuminates yellow when ACE (Advanced Concealment of Errors) is enabled. ACE protects sensitive parts of the audio frame in the presence of bit errors. The decoder must have ACE enabled for the reduction to bit errors to be effective.

#### 4.4.2.1.10 DPLL

This LED illuminates yellow when the encoder AES/EBU, SPDIF or OPTICAL digital audio input is present. An illuminated DPLL LED is required for proper operation of digital audio input signals.

#### 4.4.2.1.11 CCS

This LED illuminates when an older CCS type of compressed audio frame is transmitted. This LED should be illuminated for proper interoperation with older CCS CDQ20xx decoder to insure proper operation at all bit rates.

#### 4.4.2.1.12 MUSICAM

This LED illuminates yellow when an ISO/MPEG or new CCS compressed audio frame is transmitted. If this LED is illuminated, the cdqPRIMA will interoperate with any ISO/MPEG layer 2 compliant decoder.

#### 4.4.2.1.13 G.722

See Model 120 display for a description.

#### 4.4.2.1.14 ALG1

Currently not used.

#### 4.4.2.1.15 H.221

This LED illuminates yellow when J.52 type of H.221 multiple bonding is in effect.

375

1-28

BAD ORIGINAL



#### 4.4.2.1.16 CCSIMUX

This LED illuminates yellow when the CDQ20xx type of 2 line (2x56 or 2x64) bonding is in effect.

#### 4.4.2.2 Decoder

##### 4.4.2.2.1 PLL

See Model 120 display for a description.

##### 4.4.2.2.2 MONO

See Model 120 display for a description.

##### 4.4.2.2.3 DMONO

See Model 120 display for a description.

##### 4.4.2.2.4 STEREO

This LED illuminates yellow when an ISO/MPEG type of audio frame is received whose mode is stereo.

##### 4.4.2.2.5 JSTEREO

This LED illuminates yellow when an ISO/MPEG type of audio frame is received whose mode is stereo.

##### 4.4.2.2.6 FRAMED

See Model 120 display for a description.

##### 4.4.2.2.7 BER

See Model 120 display for a description.

##### 4.4.2.2.8 SUM

This LED illuminates when the decoder has detected any error condition. This LED is programmable and thus its meaning depends on the current definition. See the chapter entitled cdqPRIMA Logic Language

##### 4.4.2.2.9 ACE

This LED illuminates yellow when the decoder is set to expect ACE type of frame protection. ACE reduces the sensitivity of the compressed audio to bit errors. If the decoder has ACE enabled, the far end encoder must also have ACE enabled. If the

376

decoder has ACE enabled and the far end does not have ACE enabled, then the decoder will mute.

#### 4.4.2.2.10 DPLL

This LED illuminates yellow when the decoder AES/EBU, SPDIF or OPTICAL sync input is receiving a valid digital sync signal. This signal must be present if decoder digital audio synchronization is required.

#### 4.4.2.2.11 CCS

This LED illuminates when the decoder is receiving an older version of the CCS MUSICAM.

#### 4.4.2.2.12 MUSICAM

This LED illuminates yellow when the decoder is receiving either ISO/MPEG compliant frames or new CCS MUSICAM compressed digital audio frames.

#### 4.4.2.2.13 G.722

See Model 120 display for a description.

#### 4.4.2.2.14 ALG1

See Model 120 display for a description.

#### 4.4.2.2.15 H.221

This LED illuminates yellow when J.52 type of H.221 multiple bonding is in effect. The far end encoder must be utilizing J.52 type of bonding if the decoder is in the J.52 mode.

#### 4.4.2.2.16 CCSIMUX

This LED illuminates yellow when the CDQ20xx type of 2 line (2x56 or 2x64) bonding is in effect. The far end encoder must be utilizing the CDQ20xx type of 2 line bonding if the decoder is in the CDQ 2 line mode.

#### 4.4.2.3 Status

All these displays identical to the status displays on the Model 120.

### 4.4.3 Level LED's (Model 120, 220, 230)

#### 4.4.3.1 Peak & Average Level Indications

The level mode of operation allows the average level and the peak level of the signal input to the encoder and the signal output from the decoder. Each LED represents 2 dB

377

4-30

BAD ORIGINAL



of signal level and the maximum level is labeled 0 dB. This maximum level is highest level permissible at the input or at the output of the cdqPRIMA. All levels are measured relative to this maximum level. The level LED's display a 40 dB audio range.

The peak hold feature of the level LED's shows the highest level of any audio sample. This value is instantly registered and the single peak level LED moves to the value representing this signal. If the peak level of all future signals are smaller, then the peak level led slowly decays to the new peak level. The peak level LED has a fast attack and a slow decay..

#### 4.4.3.2 Stereo Image Display

The stereo image display is used to display the position of the stereo image. This is useful when setting the levels of the left and right channels to insure the proper balance.

#### 4.4.3.3 Correlation Display

This display is used to check if the left and right channels are correlated (+1). If the left and right channels are correlated, then they can be mixed to mono.

#### 4.4.3.4 Message Display

The level LED's can be used to display a scrolling message.

#### 4.4.3.5 Selective Dimming

The Status, Encoder and Decoder groups of LED's can be independently dimmed to allow emphasis of a particular group.

#### 4.4.4 Headphone Status Indicators (Model 120, 220 & 230)

The headphone indicators at the far right of the level displays are used to denote the signal output to the headphones. If both LED's are illuminated, then the left channel is output to the left earphone and the right audio channel is output to the right earphone. If only the left LED is illuminated, the left audio channel is output to both the left and right headphone. Similarly if the right channel headphone LED is illuminated.

378

BAD ORIGINAL 431



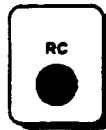
## 4.5 Front Panel Connectors



**Figure 6**  
Headphone Jack

### 4.5.1 Headphone Jack (Model 120, 220 & 230)

The front panel 1/4 inch headphone jack is located on the front panel for convenient monitoring of input or output signals. The level and control of the headphone output is controlled by the front panel push buttons under the HP heading or by remote control commands.



**Figure 7**  
Remote Control  
Port

### 4.5.2 Front Panel Remote Control Port (Model 120, 220 & 230)

The front panel remote control port is used to control all internal operations of the cdqPRIMA. It has the same functionality as the rear panel remote control connector.

## 4.6 Power Up Boot Sequence

At system power up, the cdqPRIMA loads the control processor and the various DSP's (Digital Signal Processors) from FLASH memory. It perform various power on checks and then starts execution of all its sub-systems.

379

## **4.7 System Setup**

### **4.7.1 Menu Operation**

The LCD menu sub-system is arranged like a tree. The up, right, left and enter keys allow navigation through the tree. See the chapter entitled LCD MENU TREE SUMMARY for the details of the menu tree.

### **4.7.2 Speed Dial Loading**

#### **4.7.2.1 Manual Loading**

#### **4.7.2.2 Via Remote Control**

## **4.8 Digital InterFace (DIF) Setup**

Before any connection to the outside world is possible, the digital interface modules in the cdqPRIMA must be defined. They will be set at the factory but if the Digital Interface Modules (DIM's) are rearranged, then the cdqPRIMA must be notified. This notification is done by the CIF remote control command or the Define I/F on the LCD menu.

There are 2 basic types of interfaces and these are TA (terminal adapter) and non-TA types (X.21, RS422, RS485).

There is one slot in the 1xx series models and there are 3 slots in the 2xx series. The slots are numbered

- DIF12
- DIF34
- DIF56

and are associated with digital interface 1 and 2 for DIF12 and so on.

In the future, the DIF12 slot will be expanded to include DIF34 as well.

## **4.9 Dialing with Internal ISDN Terminal Adapter(s)**

### **4.9.1 General Dialing and Auto Reconnect**

The cdqPRIMA has two methods of dialing. They are single line dialing and multiple line dialing (speed dialing). For either mode of dialing, it is possible to enable automatic reconnect. This feature allows the automatic reconnection of a dropped line. If auto reconnect is enabled (see the CAC command) when a line is dialed, then it will be reconnected if either the far end disconnected the call or the network drops the call. If the calling end drops the call, the the line will not be automatically reconnected.

380

4-33

BAD ORIGINAL

#### 4.9.2 Manual Dialing

Dialing a number with the cdqPRIMA is straightforward. Depress the DIAL button on the front panel and enter the DIF, the digital data rate (56 or 64) and the far end phone number. The left and right arrow keys are used to make the proper selection. The ENTER key is depressed to activate each selection. While the cdqPRIMA is dialing, the front panel LED (on the units with LED's) blinks for the DIF that is dialing. When the line is connected, the DIF LED illuminates with a steady light.

#### 4.9.3 Hanging Up a Connection

The cdqPRIMA allows the disconnection of individual lines or all connected lines. To initiate the disconnection process, depress the front panel button labeled **END**. Make the **ALL** or the individual line selection and depress the **ENTER** button to disconnect the line or lines.

#### 4.9.4 Speed Dial

Speed dialing requires that the speed dial configuration is entered. Assuming that this number is entered, then simply depressing the **SDIAL** button on the front panel followed by the entering of the speed dial ID and then depressing the **ENTER** button causes the far end number(s) to be dialed and the cdqPRIMA setup as required.

If speed dialing is used to establish the connection, the **END** key is used to terminate the call just like any other connection is disconnected.

#### 4.10 Resetting to Factory Defaults

When the cdqPRIMA is first turned on, it goes through a power up sequence. The first stage of the boot process is the ROM boot and the second stage is the FLASH boot. At the end of the second stage of the boot process, the cdqPRIMA looks to see if one of 4 keys are depressed. Depressing one of these keys has the following result.

- 1 reset all operational parameters (execute the CDF command)
- 2 erase all speed dial entries
- 3 set all psychoacoustic parameters to the factory default
- 0 all the above operations

The front panel button should be depressed until an acknowledgement of the depressed key is shown on the LCD screen. For example, if the 0 key is depressed during power up, then it should be held until the message

**TOTAL RESET OF DEFAULT PARMS**

appears.

381

4-34

BAD ORIGINAL

## 5. Digital Timing

The timing of the digital sections of the encoder and the decoder are controlled by various cdqPRIMA remote control commands.

The decoder timing is derived in 4 different ways. These modes of timing are set by the DTI command and are

- NORMAUTO
- INTAUTO
- INT
- AES

Let us first examine the NORMAUTO mode of timing. In this mode of operation, the timing of the decoder output is directly connected to the DA converter and the AES/EBU transmitter for the sampling rates of 48, 44.1 and 32 kHz. For the sampling rates of 24, 22.04 and 16, the decoder output is rate adapted before it goes to the DA and the

<i>Decoder Sampling Rate</i>	<i>Rate Adaption Used</i>	<i>Output Sampling Rate</i>
48	NO	48
44.1	NO	44.1
32	NO	32
24	YES	48
22.05	YES	32
16	YES	29.5

**Table 5-1**  
**Decoder rate adaption**

AES/EBU transmitter. The table below shows the configurations.

382

5-35

BAD ORIGINAL



The block diagram of the timing is shown below for the non rate adapted and the rate adapted case.

For **DTI** set to **INTAUTO**, a rate adaptor is used in all cases. The operational table for this mode is shown below.

<i>Decoder Sampling Rate</i>	<i>Rate Adaption Used</i>	<i>Output Sampling Rate</i>
48	YES	48
44.1	YES	48
32	YES	48
24	YES	48
22.05	YES	32
16	YES	29.5

**Table 5-2**  
**Decoder rate adaption**

If **DTI** is set to **INT**, then rate adaption is always used and the **DDO** command is used to set the output sampling rate. Care must be taken when utilizing the **DDO** command to set the sampling rate because not all combinations of rates are possible. See the **DDO** command for the table of possibilities.

If **DTI** is set to **AES**, then the output sampling rate is determined by the AES sync input. The decoder sync input may or may not be available. The **DES** command is used to control the timing requirement for the sync input.

### 5.1 Decoder direct connect

383

5-36

BAD ORIGINAL 

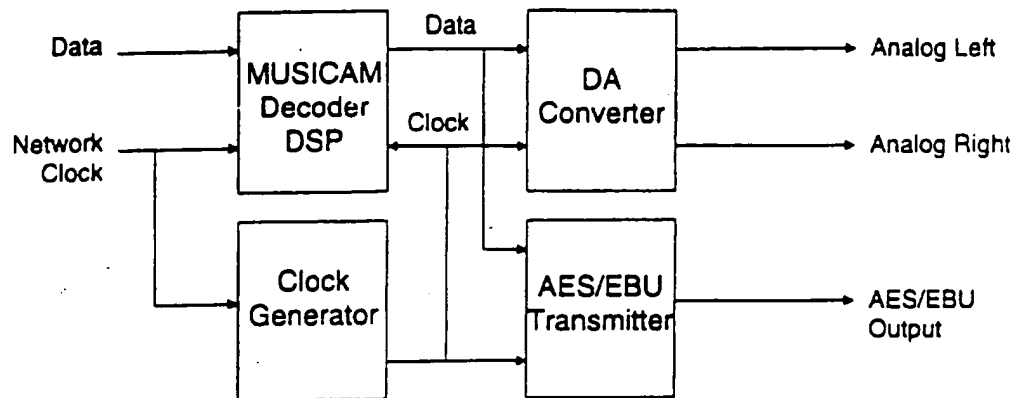
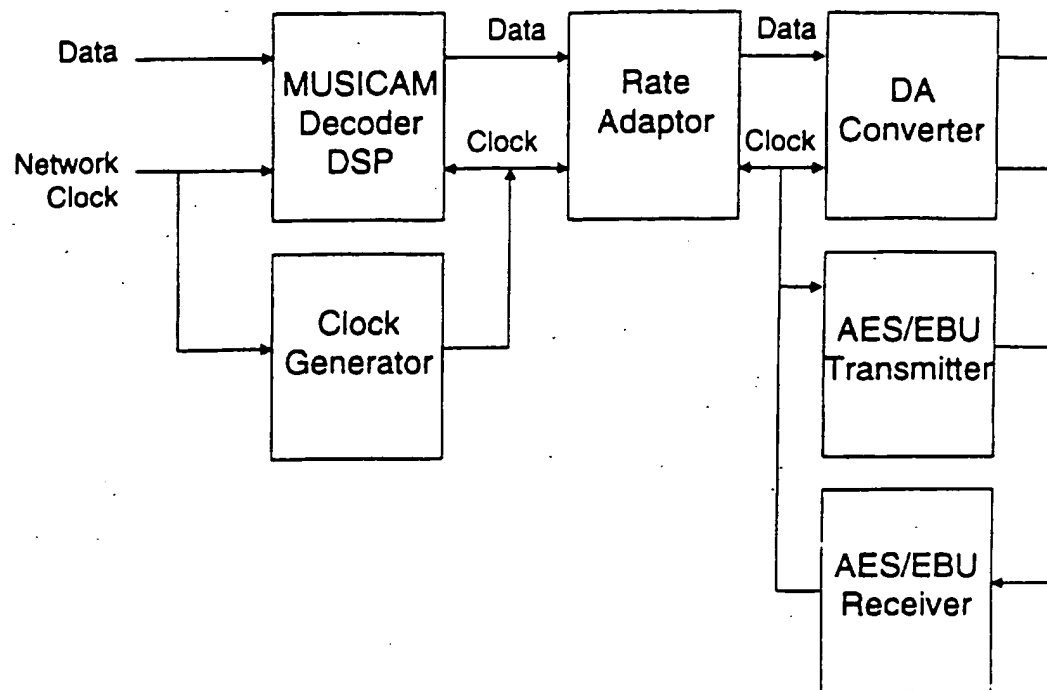


Figure 5-1

Decoder output timing with AES/EBU sync disabled or not present using normal timing

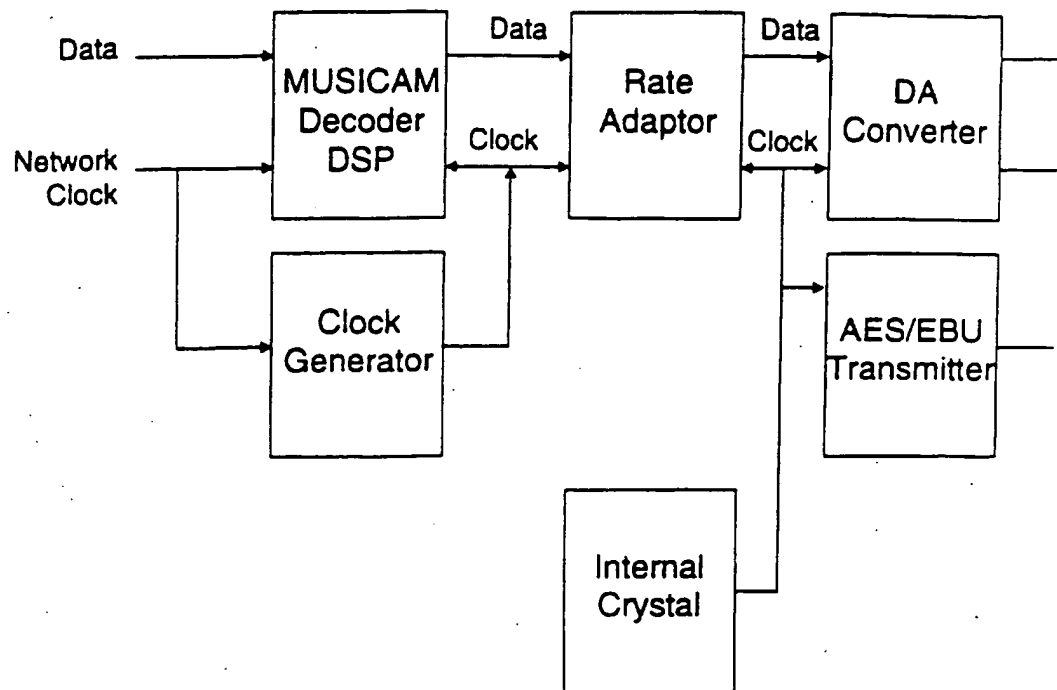
## 5.2 Decoder With Rate Adaption

**Figure 5-2**

**Decoder output timing with AES/EBU sync enabled and present using AES timing**

### **5.3 Decoder with Rate Adaption**

385

**Figure 5-3**

**Decoder output timing with AES/EBU sync disabled or not present using internal crystal timing**



## 6. Psychoacoustic Parameter Adjustment

There are 32 psychoacoustic parameters which control the cdqPRIMA. These parameters are numbered 0 through 31 and are set by the **EPP** command. Each of the 31 parameters can be of one of 4 types. These are dB, Bark, floating point and integer. The type of each parameter is set by the **EPY** command and should not be changed.

There are 20 different compressed digital audio bit rates and 6 sampling rates. This makes a total 120 different psychoacoustic parameter tables. The tables are numbered 0 to 239. The tables from 0 to 119 hold user defined parameters while the tables from 120 to 239 hold the factory defined tables. The tables from 120 to 239 should never be changed but they can be copied to a user defined table (0 to 119) and modified.

When the encoder is set to operate at a specified sampling rate and bit rate, the corresponding psychoacoustic table is loaded into the encoder. The current table number used for each sampling rate and bit rate can be displayed or changed by the **EPT** command.

To modify a factory default table, store it in a user table and tell the encoder to use the new table is done as follows.

1. Find the default table number for the desired sampling and bit rate by the **EPD** command. The number returned ranges between 120 and 239 and is the psychoacoustic table number used for the specified sampling and bit rate (called the *default table number*). Remember the second number returned by this command because it is usually used as the table number to store the modified table into (called the *suggested new table number*).
2. Execute the **EPL** command with the *default table number* to read the psychoacoustic table into memory and to download it to the encoder DSP.
3. Modify the psychoacoustic parameters with the **EPP** command until the desired audio quality is achieved.
4. Store the modified table in the *suggested new table number* by the **EPS** command.
5. Tell the encoder to use this new table for the specified sampling and bit rate by executing the **EPT** command.

It is possible and often useful to use the **EPT** command to assign multiple sampling and bit rates to the same table to minimize the table building effort.

387

(6-40)

BAD ORIGINAL



## 7. Prima Logic Language

The conversion of events, such as silence detection, into actions, such as relay closures, is handled by the cdqPRIMA Logic Language (PLL). The PLL is a simple but powerful language designed specifically for the cdqPRIMA monitor and control.

At the cdqPRIMA, there are various inputs called events. Examples of events are switch closures and silence detection. These Events are all binary in nature and are on or off (high or low). Events are mapped into Actions by Boolean logic which includes AND, OR and NOT operators. The group of Events joined by the Boolean operators is called an Event Expression. The real time values of the various events can be displayed by executing the CEV command.

Approximately every .01 seconds, each Event Expression associated with an Action is evaluated and the corresponding Action set true or false.

Silence detection Events are generated by a silence detector.

The silence detector sensitivity is determined by various parameters. For example, the level and duration of silence must be defined which causes a silence Event. See the MQC, MQD, MQL and MQT commands.

The BER detector also has parameters which must be set. See the

MBD, MBL, MBR and MBU commands.

The Actions are binary also and thus are either true or false (high or low, on or off). This means that the output Action will do something such as open or close a relay, light or extinguish a LED. A real time snapshot of the Actions can be seen by executing the CRA command. The Actions are also latched. The latched values are read by the CLA command and are cleared by the CAR command. The purpose of the latched Actions concept is to see if an Action occurred anytime in the past. This allows the detection of transient Actions (Actions which occur and then disappear).

Actions can also be the result of transitions of an Event or Event Expression from low to high or high to low. For example, the LED display might display a message when the silence detector goes from not audio present (not silent) to no audio detected (silent).

Actions perform operations at the local cdqPRIMA, such as close a relay. Actions can also be exported to a far end cdqPRIMA and can be used as input Events at the far end. There are 12 logical connections from

the near end cdqPRIMA to a far end cdqPRIMA. These connections are called links. These links are numbered from 0 to 11.

Some of the Actions described above are physical. They actually do something which can be observed. There are two other classes of

388

7-1

BAD ORIGINAL



Actions which are logical (non-physical). The first type has already been discussed and are the links between the near and far end cdqPRIMA's.

The second type of logical action is called a Virtual Action. These are also Boolean. When a Virtual Action is asserted, a cdqPRIMA Remote Control Command (PRCC) can be executed. Virtual Actions are executed when the Event Expression is high (asserted). Since Virtual Actions

are evaluated every .01 seconds, then the following PLL statement produces an unexpected result.

CEV VA0 CIO

As long as CIO is high, then once every .01 seconds, Virtual Action 0 is executed. What is probable meant by this expression is that when CIO changes from a low to a high, then Virtual Action 0 should be executed. In a discussion to follow, it will be seen that this result can be easily achieved.

Actions are named below. See Fig. 8 for reference.

RL0 = relay 0 contact closure  
 RL1 = relay 1 contact closure  
 RL2 = relay 2 contact closure  
 RL3 = relay 3 contact closure  
 RL4 = relay 4 contact closure  
 RL5 = relay 5 contact closure  
 RL6 = relay 6 contact closure  
 RL7 = relay 7 contact closure  
 SC1 = send cue LED  
 RC1 = receive cue LED  
 RLS = summary alarm relay  
 VA0 = virtual action 0  
 VA1 = virtual action 1  
 VA2 = virtual action 2  
 VA3 = virtual action 3  
 LN0 = action exported to far end cdqPRIMA on link 0  
 LN1 = action exported to far end cdqPRIMA on link 1  
 LN2 = action exported to far end cdqPRIMA on link 2  
 LN3 = action exported to far end cdqPRIMA on link 3  
 LN4 = action exported to far end cdqPRIMA on link 4  
 LN5 = action exported to far end cdqPRIMA on link 5  
 LN6 = action exported to far end cdqPRIMA on link 6  
 LN7 = action exported to far end cdqPRIMA on link 7  
 LN8 = action exported to far end cdqPRIMA on link 8  
 LN9 = action exported to far end cdqPRIMA on link 9  
 LN10 = action exported to far end cdqPRIMA on link 10  
 LN11 = action exported to far end cdqPRIMA on link 11  
 ESM = encoder summary alarm  
 DSM = decoder summary alarm

LN0..LN11 are exported to the far end cdqPRIMA while the other actions are performed only locally.

389

7-2

BAD ORIGINAL



The exported Actions are sent to the far end whenever the Action changes state or whenever a link timer has expired. This link timer interval is set by the ELU command. The result is that the exported actions are repeatedly sent to the far end even if no change has occurred. This is an attempt to communicate with the far end even in the presence of bit errors on the transmission line.

Events are named as follows.

- OI0 = optical isolator input 0
- OI1 = optical isolator input 1
- OI2 = optical isolator input 2
- OI3 = optical isolator input 3
- OI4 = optical isolator input 4
- OI5 = optical isolator input 5
- OI6 = optical isolator input 6
- OI7 = optical isolator input 7
- BER = decoder bit error detector
- OOF = decoder out of frame detector
- SEL = enc left channel silence detector
- SER = enc right channel silence detector
- SDL = dec left channel silence detector
- SDR = dec right channel silence detector
- S' = encoder stereo silence detector
- SD = decoder stereo silence detector
- CI0 = computer input 0
- CI1 = computer input 1
- CI2 = computer input 2
- CI3 = computer input 3
- CI4 = computer input 4
- CI5 = computer input 5
- CI6 = computer input 6
- CI7 = computer input 7
- DDAPLL = decoder digital audio pll
- EDAPLL = encoder digital audio pll
- TI0 = timer 0 running
- TI1 = timer 1 running
- TS0 = timer 0 just expired
- TS1 = timer 1 just expired
- EPL = encoder pll locked
- DPL = decoder pll locked
- LN0 = imported action from far end cdqPRIMA on link 0
- LN1 = imported action from far end cdqPRIMA on link 1
- LN2 = imported action from far end cdqPRIMA on link 2
- LN3 = imported action from far end cdqPRIMA on link 3
- LN4 = imported action from far end cdqPRIMA on link 4
- LN5 = imported action from far end cdqPRIMA on link 5
- LN6 = imported action from far end cdqPRIMA on link 6
- LN7 = imported action from far end cdqPRIMA on link 7
- LN8 = imported action from far end cdqPRIMA on link 8
- LN9 = imported action from far end cdqPRIMA on link 9
- LN10 = imported action from far end cdqPRIMA on link 10
- LN11 = imported action from far end cdqPRIMA on link 11
- DSPD = decoder dsp dead
- DSPE = encoder dsp dead

DSPR = reed-soloman dsp dead  
DSPV = vu meter dsp dead

The events LNO .. LN11 come from the far end cdqPRIMA.

Depressing the front panel on and off cue buttons set and clear Event C11 and C12. Front panel cue button 1 corresponds to Event C11 while button 2 corresponds to Event C12

The front panel LED's labeled

SCUE1 (SC1)

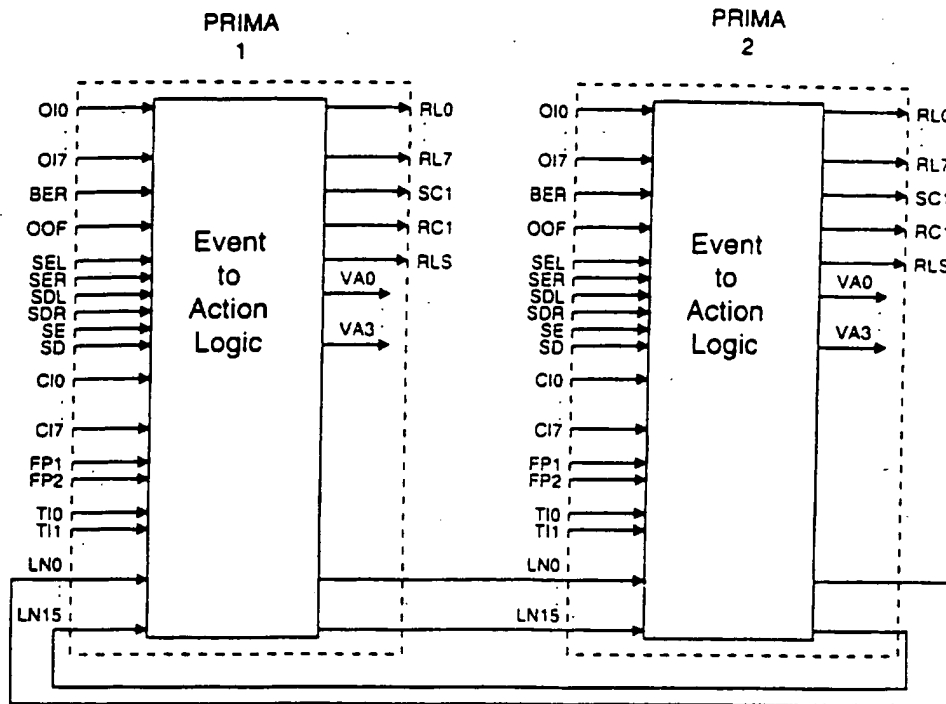
RCUE1 (RC1)

ESUM (ESM)

DSUM (DSM)

are actually defined by the PLL. They are illuminated based on the Action shown in parenthesis in the above list. This means that the state of these LED's plus all the relays are completely user definable and can be remapped to meet the needs of different applications.

The figure below shows the complete interconnection of two cdqPRIMA's with all the events and actions.

**Events**

OI0..OI7 - Optical Isolated / TTL inputs.

BER - Bit error rate detector

OOF - Out of frame detector

SEL - Encoder left channel silence detector

SER - Encoder right channel silence detector

SDL - Decoder left channel silence detector

SDR - Decoder right channel silence detector

SE - Encoder stereo silence detector

SD - Decoder stereo silence detector

CI0..CI7 - Computer simulated switch closures

FP1..FP2 - Front panel cue push buttons 1 and 2

TI0..TI1 - Timers 0 and 1

LN0..LN15 - Links from far end PRIMA

**Actions**

RL0..RL7 - Relay closures

SC1 - Send cue LED

RC1 - Receive cue LED

RLS - Summary relay closure

VA0..VA3 - Virtual actions

LN0..LN15 - Link to far end PRIMA

Figure 8  
PRIMA Monitor and Control Block Diagram

The logical operators are

& = and

# = or

! = not

The operator precedence is

! = 3

& = 2

# = 1

where 3 is the highest precedence.

An expression can have a maximum of 4 OR terms. If more than 4 OR terms are needed, a simple technique called DeMorgan's Theorem can be used to change OR's to AND's. There is no limit on AND terms.

DeMorgan's Theorem states

$$A = B \& C$$

is equivalent to

$$A = !( !B \# !C)$$

Thus the equation which contains many or terms

$$RLS = CI0 \# CI1 \# CI2 \# CI3 \# CI4 \# CI5$$

can be rewritten as

$$RLS = !( !CI0 \& !CI1 \& !CI2 \& !CI3 \& !CI4 \& !CI5 )$$

using DeMorgan's Theorem.

The + and - introduce the concept of actions based on transitions (edges). For example, if a LED message should occur when the front panel CUE 1 ON button is pressed, then the following PLL commands can be used.

CVA 0 CLM 10 HELLO WORLD

CEA VAO + (CI1)

The first statement sets Virtual Action 0 to execute the CLM 10 HELLO WORLD command. The second line states that when computer input 1 (the CUE 1 button) changes from low to high (a + edge), then Virtual Action 0 is set to a 1 (executed).

393

The LED message can be immediately suppressed when the CUE 1 OFF button is depressed if the following additional PLL statements are executed.

CVA 1 CLM 0

CEA VA1 -(CI1)

The first PLL statement associates the CLM 0 command with Virtual Action 1. The second statement activates VA1 on the high to low transition

(- edge) of the CUE 1 button.

The current PLL only allows parentheses around an entire expression.

The Actions and Events are considered identifiers.

The full use of the ( and ) operators will be allowed in future releases. See the command CDF for the default settings of the CEA commands.

Note that RI4 .. RI7 are not available on the 1xx series CODEC's but they are allowed in the PLL even though they won't do anything.

The ! operator can be used in conjunction with the + and - operator.

For example

RL0 = OI0 Relay 0 follows the level of optical input 0

RL0 = !OI0 Relay 0 follows the inverted level of optical input 0

RL0 = +OI0 Relay 0 closes when OI0 changes from a low to a high

(note that there is no way to open relay 0)

RL0 = -OI0 Relay 0 closes when OI0 changes from a high to a low

(note that there is no way to open relay 0)

RL0 = +!OI0 Relay 0 closes when OI0 changes from a low to a high

(note that there is no way to open relay 0)

Look at another simple example. Optically isolated input 2 is connected to link 5 by the command

CEA LN5 OI2

This means that when optically isolated input 2 becomes a 1, then link 5 becomes a 1 and when OI2 becomes a 0, then link 5 is 0. Remember that the Link 5 Action is exported to the far end cdqPRIMA and becomes an input Event. More on this later. For now, we will just concentrate on the language.

394





The inverted input 2 is connected to link 5 by the command

```
CEA LN5 !OI2
```

This means that when optically isolated input 2 becomes a 0, then link 5 becomes a 1 and when OI2 becomes a 1, then link 5 is 0.

The next example is slightly more complicated.

```
CEA LN7 !(OI0 # CI1)
```

This states that link 7 will be low when either OI0 is high or CI1 is high.

The real time state of the encoder stereo silence detector state can be displayed by using the received cue LED on the front panel. This is accomplished by the following command.

```
CEA RC1 SE
```

The above command is useful when trying to debug the correct settings for the silence detector. A similar trick can be used for the BER detector.

An interesting example of the power of the PLL is shown below. The audio output can be muted when the BER detector raises to a high level and decoder is framed.

```
CVA 0 DMU BOTH
```

```
CEA VA0 +BER
```

```
CVA 1 DMU NORM
```

```
CEA VA1 -BER
```

The first command attaches the DMU BOTH command to Virtual Action 0. Virtual Action is executed when the BER detector transitions from low to high. The third line attaches the "restore DA output to normal" command to Virtual Action 1. The fourth line states that when the BER detector goes from a high BER count to a low BER count, then Virtual Action 1 is executed.

To reset the received cue led back to its original definition, type

```
CEA RC1 LN8
```

A further PLL example is shown below.

```
CEA LN10 OI2 & SD # !CI3
```

In this example, link 10 is set high if EITHER OI2 and SD are high or if CI3 is low. Remember that & is higher precedence than # so the above expression could be written (if they were allowed) as follows.

```
CEA LN10 (OI2 & SD) # !CI3
```

395



Parenthesis are currently not allowed except around the entire expression. The above example would be much clearer if parenthesis were allowed.

An example of a Virtual Action is shown below

CVA 0 SD 4

CEA VA0 +SE

In this example, the CVA command assigns the operation of speed dialing entry 4 to the virtual action 0. The CEA command states that when the stereo encoder silence detector detects silence, then it sets virtual action 0 high. As just defined, the virtual action 0 would then perform the speed dial.

A last example is

CEA LN11 OI3 & OI4 & SD # CI3 & CI4 & SD # BER

It is left to the reader to decipher this command.

## 8. LCD Menu Tree Summary

The cdqPRIMA is controlled by use of the front panel keypad and the LCD display. Front panel control of the cdqPRIMA is accomplished by navigating a menu tree utilizing the MENU keys. The commands are organized into 4 main categories and these are

- Common commands
- Encoder commands
- Decoder commands
- Maintenance commands

The current position on the menu tree surrounded by square ( [ ] ) brackets while the current value of the command is enclosed in parenthesis ( ( ) ).

The table below lists the entire contents of the menu tree and a further description of the command can be found under the cdqPRIMA remote control commands which are enclosed in parenthesis. For example, more information about the LCD TA dial command can be found under the CDI remote control command.

### 8.1 Common

#### General

Password .....	CPW	Set user's password
Version .....	CVN	Print software version number
Set defaults .....	CDF	Set default parameters

#### Level LED's

Mode .....	CVU	Set level meter mode
Message .....	CLM	Display LED message
Intensity.....	CLI	Set LED display intensity

#### Head Phones

HP input .....	CHP	Set headphone audio source
Volume .....	CHV	Set headphone volume level of current device

#### TA

Dial .....	CDI	Dial TA phone number
Hangup.....	CHU	Hangup a line or lines.

#### Conn Time

LCD Dsply .....	CDC	Display TA digital interface connect time.
Dsply Time .....	CCS	Get TA digital interface connect time
Clear Time .....	CCR	Clear TA digital interface connect time
Conn Time Rst.....	CCR	Clear TA digital interface connect time

397



Auto Answer.....CAA	Set TA auto answer mode
Auto ReCon.....CAC	Set TA auto-reconnection state
Dial Time out.....CTO	Set TA dialing timeout
Connect.....CTC	Connect to TA control port
RC Protocol.....CTP	Set TA remote control protocol usage
RC Echo.....CTE	Set TA remote control command response echo

**Speed Dial**

Speed Dial.....CSD	Speed dial a number
View dir .....	View speed dial directory
Edit dir .....	Edit speed dial directory entry
Add entry.....CSE	Enter a number in the speed dial directory
Del entry.....CDS	Delete a speed dial number
Clear all.....CSC	Clear all speed dial entries

**Digital I/F**

Define I/F.....CIF	Set digital data interface type
Lp Bk Br .....	Set loopback bit rate
<b>TA</b>	
TA SPID.....CSI	Set SPID for a Terminal Adaptor
TA ID.....CLD	Set ID for a Terminal Adaptor
TA SW TYPE CSW	Set switch type
TA loopback ..CLB	Set loopback on a digital data interface

**Other**

Sys loopback.....CSL	Set system loopback
DTR/CON.....CDT	Set state of the DTR/CON line

**RP Rmt Ctl**

Set ID.....CID	Set RS485 remote control ID
Interface.....CRI	Set remote control interface type
Port baud.....CRB	Set remote control baud rate
Protocol.....CPC	Set remote control protocol usage
Echo.....CRE	Set rear panel remote control command response

echo

**FP Rmt Ctl**

FP baud.....CFB	Set set front panel remote control baud rate
FP protocol.....CFP	Set remote control protocol usage
Echo.....CFE	Set front panel remote control command response

echo

**Time Code**

Display src.....CTI	Set Time Code readout source
---------------------	------------------------------

Display spd .....	CTS	Print Time Code speed
Display last .....	CTL	Print last Time Code received
On/Off .....	CTT	Enable/disable Time Code

**Status**

RS Read .....	CRA	Print the realtime value of the action word
LS Clear .....	CAR	Clear action word latched value
LS Read .....	CLA	Print action word latched value

**PLL**

Prt evnt .....	CEV	Print event inputs
Stop timer .....	CCT	Cancel timer
Program .....	CEA	Set event to action logic

**Async Anc Data**

MUX Baud Rate ..	CMA	Set ancillary data rate for MUX
DSP Baud Rate .....	CDR	Set ancillary data rate for encoder and decoder DSP
Mux .....	CAN	Set ancillary data mode

**Sync Anc Data**

Enc bit rate .....	ESB	Set encoder synchronous bit rate
Enc clk edge .....	ESC	Set encoder synchronous clock edge
Dec bit rate .....	DSB	Set decoder synchronous bit rate
Dec clk edge .....	DSC	Set decoder synchronous clock edge

Hot Key .....	CHK	Define hot key
---------------	-----	----------------

Virtual Act .....	CVA	Define virtual action
-------------------	-----	-----------------------

## 8.2 Encoder

### General

Bit rate .....	EBR	Set encoder bit rate
Algorithm.....	EAL	Set encoder algorithm
Algorithm mode....	EAM	Set encoder algorithm mode
Line fmt.....	ELI	Set encoder digital lines format
Sample rate .....	ESR	Encoder sampling rate
Audio source.....	EAI	Set encoder audio input source
Analog bw .....	EAB	Set encoder analog bandwidth
Volume .....	EHV	Set encoder headphone volume level
Timing.....	ETI	Encoder timing
ACE .....	ESP	Set encoder scale factor protection
Calibrate AD.....	EAD	Calibrate AD converter

### ISO Header

Copyright .....	ECR	Set encoder copyright bit in header
Emphasis.....	EEP	Set encoder emphasis bit in header
Original .....	EOR	Set encoder original bit in header
Protection .....	EPR	Set encoder protection bit in header
Private .....	EPI	Set encoder private bit in header

### Contacts

Set Switch.....	ESW	Set a simulated switch
-----------------	-----	------------------------

### Psycho

Set Parm.....	EPP	Set psychoacoustic parameter
Reset .....	EPB	Load all default psychoacoustic parameters
Tbl Num.....	EPD	Get default psychoacoustic parameter table number
Load Tbl.....	EPL	Load psychoacoustic parameter table from flash
Store Tbl .....	EPS	Store psychoacoustic parameter table in flash
Assign Tbl.....	EPT	Assign psychoacoustic parameter table

### 8.3 Decoder

#### General

Bit rate .....	DBR	Set decoder bit rate
Independent.....	DIN	Set decoder - encoder interaction
Output SR .....	DDO	Set digital output sampling rate.
Line fmt.....	DLI	Set decoder digital lines format
Timing.....	DES	Enable decoder AES sync timing
Decoding mode .....	DCO	Set decoder decoding mode
ACE .....	DSP	Scale factor protection
Calibrate DA .....	DDA	Calibrate DA converter
Algorithm.....	DAL	Set decoder algorithm
Status bits.....	DRS	Display real time status

#### Audio out

Mute .....	DMU	Mute decoder output channels
Copy/Swap.....	DCS	Set channel copy/swap mode
Test tones .....	DMD	Set decoder maintenance diagnostic mode

## 8.4 Maintenance

### Silence Det

Time left.....MQC Display quiet detector level time left  
 Set lvl.....MQL Set quiet detector level  
 Set time.....MQT Set quiet time duration  
 Read lvl.....MQD Display quiet detector level

### Peak Det

Peak lvl.....MPD Display peak detector level

### BER Det

Dsply Cnt.....MBC Display BER counter  
 Reset Cnt.....MBR Reset BER counter  
 Set Thresh.....MBL Set BER count rate limit  
 Up Cnt.....MBU Set BER up count rate  
 Down Cnt.....MBD Set BER down count rate

### OOF Det

Dsply Cnt.....MOC Display OOF counter  
 Reset Cnt.....MOR Reset OOF counter  
 Set Thresh.....MOL Set OOF count rate limit  
 Up Cnt.....MOU Set OOF up count rate  
 Down Cnt.....MOD Set OOF down count rate

### Graphic Tests

Graphics.....MTM Perform a test measurement

### PRIMA Tests

En/Dis Tests.....MET Enable hardware tests  
 Hrdwre Tests.....MHT Perform hardware tests

### Debug

Watch Port.....MWP Set watch port

### Status

Version Num.....MVN Print software version number

### Soft Dnld

Boot ROM.....MBM Boot the cdqPRIMA from ROM  
 FE Boot ROM.....MRM Boot the far end cdqPRIMA from ROM  
 RP RC Source.....MRS Set rear panel remotc control uart source

BBM Sync MSY Synchronize RAM and BBM

402



## 9. cdqPRIMA Remote Control Commands

### CAA Set TA auto answer mode

This command is used to set a digital interface TA to auto answer. It does this by asserting the DTR line on the Terminal Adaptor (TA). This command can also be used to hangup a connected call. If auto answer is set to NO, then a connected call (if any) is disconnected.

Once a digital interface line is set to no auto-answer, then it will not receive any calls. If a call is pending, and auto-answer is enabled, the the call will be answered.

See the CAD, CCR, CCS, CDI, CHU, CLD, CSI, CTC and CTO commands

**CAA di ?** print auto answer status for digital interface di

**CAA di aa** set auto answer status for digital interface di to aa

di = 1, 2, ... 6  
aa = YES or NO

### CAC Set TA auto-reconnection state

This command is used to set the TA auto-reconnection status. If ad is set to YES, then if a TA connection is dropped, it will automatically be re-established.

See the ?? commands.

**CAC ?** print TA auto-reconnection state

**CAC ad** set TA auto-reconnection state to ad

ad = YES or NO

### CAN Set ancillary data mode

This command is used to set the ancillary data mux/demux configuration. See Fig 1 for a description of the various ancillary data configuration configurations.

See the CDR, DSB and ESB commands.

**CAN ?** print current ancillary data configuration

**CAN an** set ancillary data configuration to an

an = 0, 1, ... 6

### CAR Clear action word latched value

This command clears the latched value of the action word.

The action outputs are printed as a hex number with the msb at the left and the lsb at the right. The meaning of the bits are as follows.

If a bit is high in the action word, then the corresponding action is also high.

BIT 0 RL0 - relay 0  
 BIT 1 RL1 - relay 1  
 BIT 2 RL2 - relay 2  
 BIT 3 RL3 - relay 3  
 BIT 4 RL4 - relay 4  
 BIT 5 RL5 - relay 5  
 BIT 6 RL6 - relay 6  
 BIT 7 RL7 - relay 7  
 BIT 8 SC1 - send cue 1 LED  
 BIT 9 RC1 - receive cue 1 LED  
 BIT 10 RLS - summary relay  
 BIT 11 VA0 - virtual action 0  
 BIT 12 VA1 - virtual action 1  
 BIT 13 VA2 - virtual action 2  
 BIT 14 VA3 - virtual action 3  
 BIT 15 unused  
 BIT 16 LN0 - link to far end PRIMA 0  
 BIT 17 LN1 - link to far end PRIMA 1  
 BIT 18 LN2 - link to far end PRIMA 2  
 BIT 19 LN3 - link to far end PRIMA 3  
 BIT 20 LN4 - link to far end PRIMA 4  
 BIT 21 LN5 - link to far end PRIMA 5  
 BIT 22 LN6 - link to far end PRIMA 6  
 BIT 23 LN7 - link to far end PRIMA 7  
 BIT 24 LN8 - link to far end PRIMA 8  
 BIT 25 LN9 - link to far end PRIMA 9  
 BIT 26 LN10 - link to far end PRIMA 10  
 BIT 27 LN11 - link to far end PRIMA 11  
 BIT 28 ESM - encoder summary alarm  
 BIT 29 DSM - decoder summary alarm  
 BIT 30 unused  
 BIT 31 unused

See the CEV, CEA, CLA, CRA, ELU and ESW commands.

**CAR** clear the latched value of the action word

### **CBR Set loopback bit rate**

This command is used to set the digital audio bit rate when the PRIMA is in loopback.

See the CLB and CSL commands.

**CBR ?** print loop back bit rate

**CBR lr** set loop back bit rate to lr

404

lr = 24, 32, 48, 56, 64, 96, 112,  
128, 192, 224, 256, 384

### CCR Clear TA digital interface connect time

This command is used to clear the number of seconds a terminal adaptor type of digital interface is connected.

The time connected can be read by the CCS command.

When the CCR command is executed, it clears the time in the timer.

See the CAA, CAD, CCS, CDI, CHU, CLD, CSI, CTC and CTO commands

**CCR di** clear the time in seconds a terminal adaptor is connected

di = 1, 2, ... 6

### CCS Get TA digital interface connect time

This command is used to get the number of seconds a terminal adaptor type of digital interface is connected. When the TA enters the connect state, a timer for that digital interface is started and counts seconds. When the line is disconnected, the timer is stopped but not cleared.

The time line was connected is displayed by this command. If the digital interface TA is currently connected, this command will report the current elapsed connect time.

The timer can be set to 0 by issuing the CCR command.

This command is useful for monitoring the time a call is in progress.

See the CAA, CAD, CCR, CDI, CHU, CLD, CSI, CTC and CTO commands

**CCS di** print the time in seconds a terminal adaptor is connected

di = 1, 2, ... 6

### CCT Cancel timer

This command is used to cancel an internal timer. This timer is used by the PRIMA Logic Language (PLL) to generate events which occur at some future time.

A timer cancelled by this command does not create any action.

See the CTM and CEA commands.

**CCT tn** cancel timer tn

tn = 0 or 1

405

### CDC Display TA digital interface connect time.

This command is used to display the number of seconds a terminal adaptor type of digital interface is connected. When the display is requested, then it is displayed on the LCD screen. It may cover up part of another display.

When the TA enters the connect state, a timer for that digital interface is started and counts seconds is displayed. When the line is disconnected, the timer is stopped but not cleared and it is still displayed.

When the digital interface is again connected, the timer is reset and begins counting again.

There are two forms for the command.

CDC NO

CDC YES di

The first form is used to set to inhibit the display while the second form is used to display the connect time on digital interface di.

The CDC ? command has two possible responses. They are

NO

YES di

In the first case, no digital interface connect time is displayed. In the second case the connect time for DIF di is being displayed on the LCD display.

This command is useful for monitoring the time a call is in progress.

See the CAA, CAD, CCR, CDI, CHU, CLD, CSI, CTC and CTO commands

CDC ?        print the TA connect time display status

CDC NO       stop printing the connect time on lcd display

CDC YES di   display TA digital interface connect time on DIF di

di        =        1, 2, ... 6

### CDF Set default parameters

This command is used to set the CODEC to the factory default values.

The default values are as follows:

CAA 1 YES	set auto answer on for line 1
CAA 2 YES	set auto answer on for line 2
CAA 3 YES	set auto answer on for line 3

406

9-4

BAD ORIGINAL



CAA 4 YES	set auto answer on for line 4
CAA 5 YES	set auto answer on for line 5
CAA 6 YES	set auto answer on for line 6
CAN 2	set ancillary data port to configuration 2 (mux)
CBR 128	set loopback digital interface bit rate
CDC NO	don't display connect time
CDR 9600	set decoder dsp ancillary data rate
CEA LN0 !OI0	set default actions
CEA LN1 !OI1	
CEA LN2 !OI2	
CEA LN3 !OI3	
CEA LN4 !OI4	
CEA LN5 !OI5	
CEA LN6 !OI6	
CEA LN7 !OI7	
CEA LN8 CI1	
CEA LN9 CI2	
CEA LN10 BER	
CEA LN11 OOF	
CEA ESM !EPL	
CEA DSM !DPL # BER # OOF	
CEA RLS !EPL # !DPL # BER # OOF	
CEA RL0 LN0	
CEA RL1 LN1	
CEA RL2 LN2	
CEA RL3 LN3	
CEA RL4 LN4	
CEA RL5 LN5	
CEA RL6 LN6	
CEA RL7 LN7	
CEA SC1 CI1	
CEA RC1 LN8	
CEA VA0	
CEA VA1	
CEA VA2	
CEA VA3	
CFB 9600	set front panel remote control baud rate
CFP NO	set no front panel remote control protocol
CHK 1	set to nothing in the hot hey
CHK 2	set to nothing in the hot hey
CHK 3	set to nothing in the hot hey
CHK 4	set to nothing in the hot hey
CHK 5	set to nothing in the hot hey
CHK 6	set to nothing in the hot hey
CHK 7	set to nothing in the hot hey
CHK 8	set to nothing in the hot hey
CHP E	set headphone to encoder
CID 0	set to RS485 id 0
CIF 1 NONE	set to no digital interface
CIF 2 NONE	set to no digital interface
CIF 3 NONE	set to no digital interface
CIF 4 NONE	set to no digital interface
CIF 5 NONE	set to no digital interface
CIF 6 NONE	set to no digital interface
CLB 1 NORM	set no digital loopback on DIF 1

407

CLB 2 NORM	set no digital loopback on DIF 2
CLB 3 NORM	set no digital loopback on DIF 3
CLB 4 NORM	set no digital loopback on DIF 4
CLB 5 NORM	set no digital loopback on DIF 5
CLB 6 NORM	set no digital loopback on DIF 6
CLI STATUS 10	set status led intensity
CLI ENCODER 10	set encoder led intensity
CLI DECODER 10	set decoder led intensity
CMA 2400	set mux ancillary data baud rate
CPC NO	no protocol for remote communications
CRB 9600	set remote control baud rate
CRI 232	RS232 for remote communication
CSL NORM	no system loopback
CTC NONE	no connection to any TA
CTI NONE	no time code display
CTT OFF	no time code hardware
CTO 15	set TA dialing timeout in seconds
CVA 0	set virtual action 0 to empty
CVA 1	set virtual action 1 to empty
CVA 2	set virtual action 2 to empty
CVA 3	set virtual action 3 to empty
CVU LEVEL	set level meter to level (vu mode)
DAL MPEG L2	set to MPEG layer 2
DCO ISOCCS	set decoder decoding mode to ISO and CCS
DCS NONE	set decoder output to no copy or swap
DDO 48	set to 48 khz digital output
DES NOTREQ	set decoder sync timing not required
DHV 75	set decoder headphone volume
DIN NO	set decoder to operate together with encoder
DLI L1	set to no decoder line usage
DMD NORM	set decoder maintenance diagnostic mode to normal
DMU NONE	set decoder mute to none
DSB NONE	set no decoder synchronous ancillary data
DSP NO	set to no decoder scale factor protection
DTI NORM AUTO	set decoder timing to normal
EAL MPEG L2	set to MPEG layer 2
EBR 128	set encoder to 128k bit rate
ECR NO	set no copyright bit
EEP NO	set no emphasis bit
EHV 75	set encoder headphone volume
ELI L1	set to no encoder line usage
ELU 1	set to link messages every .1 sec
EOR NO	set no original bit
EPI OFF	set no privacy bit off
EPR YES	set protection bit
EPY 0 1	set psychoacoustic parameter type
EPY 1 2	set psychoacoustic parameter type
EPY 2 1	set psychoacoustic parameter type
EPY 3 2	set psychoacoustic parameter type
EPY 4 1	set psychoacoustic parameter type
EPY 5 3	set psychoacoustic parameter type
EPY 6 1	set psychoacoustic parameter type
EPY 7 1	set psychoacoustic parameter type
EPY 8 1	set psychoacoustic parameter type
EPY 9 3	set psychoacoustic parameter type

EPY 10 4	set psychoacoustic parameter type
EPY 11 3	set psychoacoustic parameter type
EPY 12 4	set psychoacoustic parameter type
EPY 13 3	set psychoacoustic parameter type
EPY 14 1	set psychoacoustic parameter type
EPY 15 3	set psychoacoustic parameter type
EPY 16 4	set psychoacoustic parameter type
EPY 17 3	set psychoacoustic parameter type
EPY 18 4	set psychoacoustic parameter type
EPY 19 4	set psychoacoustic parameter type
EPY 20 3	set psychoacoustic parameter type
EPY 21 3	set psychoacoustic parameter type
EPY 22 1	set psychoacoustic parameter type
EPY 23 1	set psychoacoustic parameter type
EPY 24 1	set psychoacoustic parameter type
EPY 25 1	set psychoacoustic parameter type
EPY 26 4	set psychoacoustic parameter type
EPY 27 3	set psychoacoustic parameter type
EPY 28 4	set psychoacoustic parameter type
EPY 29 4	set psychoacoustic parameter type
EPY 30 1	set psychoacoustic parameter type
EPY 31 1	set psychoacoustic parameter type
ESB NONE	set no encoder synchronous ancillary data
ESP NO	set to no encoder scale factor protection
ESR 48	set encoder sampling rate to 48
ESW C10 OFF	set simulated switch 0 open
ESW C11 OFF	set simulated switch 1 open
ESW C12 OFF	set simulated switch 2 open
ESW C13 OFF	set simulated switch 3 open
ESW C14 OFF	set simulated switch 4 open
ESW C15 OFF	set simulated switch 5 open
ESW C16 OFF	set simulated switch 6 open
ESW C17 OFF	set simulated switch 7 open
ETI NORM	set encoder timing to normal
MBD 1	set BER count down counter
MBL 1	set BER limit to 1
MBR	clear the BER counter
MBU 1	set BER count up counter
MOD 1	set OOF down counter
MOL 10	set OOF limit
MOU 2	set OOF up counter
ML EL -60	set encoder left quiet threshold level
ML ER -60	set encoder right quiet threshold level
ML DL -60	set decoder left quiet threshold level
ML DR -60	set decoder right quiet threshold level
ML E -60	set encoder and decoder right quiet threshold level
ML D -60	set encoder and decoder right quiet threshold level
MLT EL 10	set encoder left quiet threshold time
MLT ER 10	set encoder right quiet threshold time
MLT DL 10	set decoder left quiet threshold time
MLT DR 10	set decoder right quiet threshold time
MLT E 10	set encoder quiet threshold time
MLT D 10	set decoder quiet threshold time
MRS RP	set rear panel remote control source to rear panel
MWP NONE	set to no watch port

**CDP** sets the defaults into the cdqPRIMA

### CDI Dial TA phone number

This command is used to dial a phone number on a specific digital interface. This command is used to set up a phone call and is primarily intended for testing. It can be used for setting up the lines individually.

To hangup a dialed line, use the CHU command.

See the CAA, CAD, CCR, CCS, CHU, CLD, CSI, CTC and CTO commands

<b>CDI di db dn</b>	dial phone number dn on digital interface di at bit rate db
<b>di</b>	= 1, 2, ... 6
<b>db</b>	= 56 or 64
<b>dn</b>	= 20 digit phone number

### CDR Set ancillary data rate for encoder and decoder DSP

This command is used to set the ancillary data rate for the encoder and decoder DSP. The control processor are also involved in the ancillary data process. This data rate is used for communications from the mux to the encoder DSP and from the decoder DSP to the de-mux.

See the CAN, DSB and ESB commands.

<b>CDR ?</b>	print encoder and decoder DSP ancillary data rate
<b>CDR dr</b>	set encoder and decoder DSP ancillary data rate
<b>dr</b>	= 300, 1200, 2400, 4800, 9600 and 38400

### CDS Delete a speed dial number

This command is used to delete a speed dial number.

See the CSC, CSD, CSE, CSF and CSN commands.

<b>CDS sn</b>	delete speed dial number sn
<b>sn</b>	= 0..255

### CDT Set state of the DTR/CON line

4/0



This command is used to set the state of the DTR/CON line on non-TA type of interfaces.

See the CIF command.

**CDT di ?** prints the state of the DTR/CON line on digital interface di

**CDT di st** set the state of DTR/CON line on digital interface di to st

di = 1, 2, ... 6

st = H or L

### CEA Set event to action logic

This command is used to set the event to action logic. See section ?? for more details about this command.

See the CAR, CCT, CEV, CLA, CTM, CRA, CVA, ELU, ESW, MBD, MBL, MBR, MBU MQC, MQD, MQL and MPT commands.

**CEA If ?** print event to link connection for link ln

**CEA If [el]** set event el to action If connection

If = LN0 .. LN11, RL0..RL7, SC1, RC1, RLS, VA0..VA3  
(, ), +, - and !'s

el = optional event logic

### CEV Print event inputs

This routine is used so print the compiled program for an action.

The event inputs are printed as a hex number with the msb at the left and the lsb at the right. The meaning of the bits are as follows.

If a bit is high in the event word, then the corresponding event is also high.

BIT 0	OI0 - optical isolator input 0
BIT 1	OI1 - optical isolator input 1
BIT 2	OI2 - optical isolator input 2
BIT 3	OI3 - optical isolator input 3
BIT 4	OI4 - optical isolator input 4
BIT 5	OI5 - optical isolator input 5
BIT 6	OI6 - optical isolator input 6
BIT 7	OI7 - optical isolator input 7
BIT 8	BER - decoder bit error detector
BIT 9	OOF - decoder out of frame detector
BIT 10	SEL - enc left channel silence detector
BIT 11	SER - enc right channel silence detector
BIT 12	SDL - dec left channel silence detector

411



BIT 13	SDR - dec right channel silence detector
BIT 14	SE - encoder stereo silence detector
BIT 15	SD - decoder stereo silence detector
BIT 16	CI0 - optical isolator input 0
BIT 17	CI1 - optical isolator input 1
BIT 18	CI2 - optical isolator input 2
BIT 19	CI3 - optical isolator input 3
BIT 20	CI4 - optical isolator input 4
BIT 21	CI5 - optical isolator input 5
BIT 22	CI6 - optical isolator input 6
BIT 23	CI7 - optical isolator input 7
BIT 24	DDAPLL - decoder digital audio pll locked
BIT 25	EDAPLL - encoder digital audio pll locked
BIT 26	TIO - timer 0
BIT 27	TII - timer 1
BIT 28	TS0 - timer 0 stopped
BIT 29	TS1 - timer 1 stopped
BIT 30	EPL - encoder phase locked loop
BIT 31	DPL - decoder phase locked loop
BIT 32	LN0 - decoder link 0
BIT 33	LN1 - decoder link 1
BIT 34	LN2 - decoder link 2
BIT 35	LN3 - decoder link 3
BIT 36	LN4 - decoder link 4
BIT 37	LN5 - decoder link 5
BIT 38	LN6 - decoder link 6
BIT 39	LN7 - decoder link 7
BIT 40	LN8 - decoder link 8
BIT 41	LN9 - decoder link 9
BIT 42	LN10 - decoder link 10
BIT 43	LN11 - decoder link 11
BIT 44	DSPD - decoder dsp dead
BIT 45	DSPE - encoder dsp dead
BIT 46	DSPR - reed-soloman dsp dead
BIT 47	DSPV - vu dsp dead
BIT 48	FRAMED - decoder dsp framed

See the CEA, CAR, CLA, CRA, ELU and ESW commands.

**CEV ev**      print event inputs ev

ev      =      ALL,

OIO..OI7, BER, OOF, SEL, SER,  
 SDL, SDR, SE, SD, CI0..CI7,  
 DDAPLL, EDAPLL,  
 TIO, TII, TS0, TS1, LN0..LN11,  
 ESM, DSM, EPL,  
 DPL, DSPD, DSPE, DSPR, DSPV,  
 FRAMED

### CFB Set set front panel remote control baud rate

This command is used to set the front panel remote control baud rate.

412

The baud rate can be 1200, 2400, 4800, 9600 or 38400 baud.

See the CFE and CFP commands.

<b>CFB ?</b>	print front panel remote control baud rate
<b>CFB fb</b>	set front panel remote control baud rate to fb
<b>fb</b>	= 1200, 2400, 4800, 9600 or 38400

#### **CFE Set front panel remote control command response echo**

This command is used to set the front panel remote control command echo. When downloading new software in flash, it is advisable to turn off command echo to speed the download process.

See the CFB and CFP commands.

<b>CFE ?</b>	print front panel remote control command response echo state
<b>CFE re</b>	set front panel remote control command response echo state to re
<b>re</b>	= YES or NO

#### **CFP Set remote control protocol usage**

This command forces the encoder to utilize a protocol on all front panel remote control messages. If no protocol is used, then point to point communications is assumed (a pc is connected to only 1 encoder). If protocol is used, then each CODEC device must have an id set by the CID command. The protocol can then select the specified device. Protocol communication can be used for point to point and point to multipoint communication.

If protocol was not enabled and it is enabled, the response will be in protocol mode (even though the input command was not in protocol mode) with a BSN of 0.

See the CFB and CFE commands.

<b>CFP ?</b>	print remote control protocol mode
<b>CFP fp</b>	set remote control protocol mode to fp
<b>fp</b>	= YES or NO

#### **CHK Define hot key**

This command is used to define a hot key. A hot key is front panel push button f1 to f8 which, when pushed, activates a command. For example

**CHK 2 CSD 5**

assigns the PRIMA Remote Control Command

**CSD 5**

to hot key 5 2.

**CHK hk ?** print command associated with hot key hk

**CHK hk cm** attach command cm to hot key hk

**hk** = 1 .. 8

**cm** = any cdqPRIMA Remote Control Command (PRCC)

### **CHP Set headphone audio source**

This command is used to set the headphone audio source.

The possibilities are the encoder (E, EL or ER), decoder (D, DL or DR) or mute (M).

For both the encoder and the decoder, there exist the possibilities of both channels (E or D), left channel only (EL or DL) or right channel only (ER or DR).

See the CHV, DHV and EHV commands.

**CHP ?** print headphone audio source

**CHP hp** set headphone audio source to hp

**hp** = E, EL, ER, D, DL, DR or M

### **CHU Hangup a line or lines.**

This command is used to hang up a connected line. It can only be used on digital interface lines designated as TA's.

The command to connect a line is CDI. To connect multiple lines, use the CAD command.

See the CAA, CAD, CCR, CCS, CDI, CLD, CSI, CTC and CTO commands

**CHU df** hangup a line or lines.

**df** = ALL, 1, 2, ... 6

### **CHV Set headphone volume level of current device**

4/4

This command is used to set the volume level of the currently selected device (encoder or decoder). The level applies to the selected device and controls the level of the audio output to the headphone jack.

See the CHP, DHV and EHV commands.

**CHV ?** print headphone volume level of currently selected device

**CHV hv** set decoder headphone volume to hv

hv = 0 .. 127, + or -

#### **CID Set RS485 remote control ID**

This command is used to set the RS485 id of the CODEC. This ID is used by remote control software to address the CODEC in an RS485 environment.

**CID ?** prints the RS485 ID

**CID id** set the RS485 ID to id

id = 0, 1, ... 30

#### **CIF Set digital data interface type**

This command is used to set the type of digital data interface. For the cdqPRIMA 2xx series, the interfaces are numbered from 1 through 6.

On the cdqPRIMA 1xx series, the interfaces are numbered 1 and 2.

If the interface is set to a TA, then auto answer is turned on.

If the interface type is X.21, V.35 or RS422, then the line state is set to CONNECTED (the CST command displays the connection status) permanently.

If the interface type is a type of TA, then the connection state is set to CONNECTED once the connection has been established.

See the CDT command.

**CIF di ?** prints the interface type for digital interface di

**CIF di it** set digital interface di to it

di = 1, 2, ... 6

it = TA101, TA201, TA202, X.21, V.35, RS422 or NONE

#### **CLA Print action word latched value**

This command prints the latched value of the action word.

415

9-13

BAD ORIGINAL

If a specific bit is requested, then the resulting value is 0 or 1. If all is specified, the the entire action word is printed in hex with the right most bit (LSB) corresponding to BIT 0.

The action outputs are printed as a hex number with the msb at the left and the lsb at the right. The meaning of the bits are as follows.

If a bit is high in the action word, then the corresponding action is also high.

BIT 0 LN0 - link to far end PRIMA 0  
 BIT 1 LN1 - link to far end PRIMA 1  
 BIT 2 LN2 - link to far end PRIMA 2  
 BIT 3 LN3 - link to far end PRIMA 3  
 BIT 4 LN4 - link to far end PRIMA 4  
 BIT 5 LN5 - link to far end PRIMA 5  
 BIT 6 LN6 - link to far end PRIMA 6  
 BIT 7 LN7 - link to far end PRIMA 7  
 BIT 8 LN8 - link to far end PRIMA 8  
 BIT 9 LN9 - link to far end PRIMA 9  
 BIT 10 LN10 - link to far end PRIMA 10  
 BIT 11 LN11 - link to far end PRIMA 11  
 BIT 12 ESUM - encoder summary alarm  
 BIT 13 DSUM - encoder summary alarm  
 BIT 14 unused  
 BIT 15 unused  
 BIT 16 RL0 - relay 0  
 BIT 17 RL1 - relay 1  
 BIT 18 RL2 - relay 2  
 BIT 19 RL3 - relay 3  
 BIT 20 RL4 - relay 4  
 BIT 21 RL5 - relay 5  
 BIT 22 RL6 - relay 6  
 BIT 23 RL7 - relay 7  
 BIT 24 SC1 - send cue 1 LED  
 BIT 25 RC1 - receive cue 1 LED  
 BIT 26 RLS - summary relay  
 BIT 27 VA0 - virtual action 0  
 BIT 28 VA1 - virtual action 1  
 BIT 29 VA2 - virtual action 2  
 BIT 30 VA3 - virtual action 3  
 BIT 31 not used

See the CEV, CEA, CAR, CRA, ELU and ESW commands.

**CLA aw** print latched value bit aw of the action word

aw = ALL, LN0..LN11, ESM, DSM,  
 RL0..RL7,  
 SC1, RC1, RLS, VA0..VA3

### CLB Set loopback on a digital data interface

This command is used to set a loopback at the digital interface whose

number is given by ta.

For the cdqPRIMA, the interfaces are numbered from 1 through 6.

In the LB state, any data sent to the digital interface by the encoder is "looped back" to the decoder.

The CLB type of loopback is performed at the digital interface such as the TA. The CSL loopback is performed before the signals reach the digital interface.

See the CBR and CSL commands.

**CLB di ?** prints the digital interface type

**CLB di lb** set loopback state lb on digital interface di

di = 1, 2, ... 6

lb = LB or NORM

#### **CLD Set ID for a Terminal Adaptor**

This command is used to set the ID for the digital interface.

The ID is only used for TA's in North America.

See the CAA, CAD, CCR, CCS, CDI, CHU, CSI, CTC and CTO commands

**CLD di ?** prints the ID for digital interface di

**CLD di ld** set ID for digital interface di to ld

di = 1, 2, ... 6

ld = 20 digit number

#### **CLI Set LED display intensity**

This command is used to set the intensity of the LED display.

The intensity can range from 0 to 15 where 15 is the brightest intensity.

This command

**CLI gr ?** print current intensity

**CLI gr iy** set LED group gr to intensity iy

gr = STATUS, ENCODER and DECODER

iy = 0..15

417

**CLM Display LED message**

This command is used to display a message on the LED screen. It cancels an existing message on the LED screen.

For example

CLM 2 5 MIN TO AIR

displays the message 5 MIN TO AIR for 2 seconds.

Another example

CLM 20 BER OCCURRED

displays the message BER OCCURRED for 20 seconds.

The command

CLM 0

terminates the display of any message on the LED and returns the display to vu mode.

**CLM du [ms]** displays message **ms** for duration **du**

**du** = 0 .. message duration in seconds

**ms** = any ascii message up to 30 characters

**CMA Set MUX ancillary data baud rate**

This command is used to set the MUX asynchronous ancillary baud rate. The MUX ancillary data rate is different from the DSP ancillary data baud rate.

See the CDR command.

**CMA ?** print MUX ancillary baud rate

**CMA ma** set mux ancillary data baud rate to **ma**

**ma** = 300, 1200, 2400, 4800, 9600 or 19200

**COM Comment command**

This command takes an arbitrary number of arguments and ignores them. It is intended to be used for comments in a batch file.

**COM x1 x2 x3 ..** comment command (no operation)

418



x1 = any set of characters

x2 = any set of characters

### CPC Set remote control protocol usage

This command forces the encoder to utilize a protocol on all remote control messages. If no protocol is used, then point to point communications is assumed (a pc is connected to only 1 encoder). If protocol is used, then each CODEC device must have an id set by the CID command. The protocol can then select the specified device. Protocol communication can be used for point to point and point to multipoint communication.

If protocol was not enabled and it is enabled, the response will be in protocol mode (even though the input command was not in protocol mode) with a BSN of 0.

If the RS-485 remote control interface is selected via the CRI command, then multiple CODEC's (up to 30) can be on the bus.

**CPC ?** print remote control protocol mode

**CPC pc** set remote control protocol mode to pc

pc = YES or NO

### CPW Set user's password

This command allows the user to enter a password, thus raising his/her security level

**CPW?** prints the previous password

**CPW pw** enter the next password pw

pw = a big decimal number

### CQQ Print command summary for common commands

This command is used to print a summary of all the Cxx commands.

See the DQQ, EQQ, MQQ and QQQ (HELP) commands.

**CQQ** print command summary

### CRA Print the realtime value of the action word

419

This command prints the realtime value of the action word.

If a specific bit is requested, then the resulting value is 0 or 1. If all is specified, the entire action word is printed in hex with the right most bit corresponding to BIT 0.

See CLA for definition of the hex representation of the action word.

See the CEV, CEA, CAR, CRA, ELU and ESW commands.

**CRA aw**      print realtime value bit **aw** of the action word.  
**aw**      =      ALL, LN0..LN11, RL0..RL7,  
                 SC1, RC1, RLS, VA0..VA3, ESM, DSM

### **CRB Set remote control baud rate**

This command is used to set the remote control interface baud rate.

The baud rate can be 1200, 2400, 4800, 9600 or 38400 baud.

**CRB ?**      print remote control baud rate  
**CRB rb**      set remote control baud rate to **rb**  
**rb**      =      1200, 2400, 4800, 9600 or 38400

### **CRE Set rear panel remote control command response echo**

This command is used to set the rear panel remote control command echo. When downloading new software in flash, it is advisable to turn off command echo to speed the download process.

See the CRB command.

**CRE ?**      print rear panel remote control command echo state  
**CRE re**      set rear panel remote control command echo state to **re**  
**re**      =      YES or NO

### **CRI Set remote control interface type**

This command is used to set the remote control interface type to RS232 or RS485.

**CRI ?**      print remote control input source  
**CRI ri**      set remote control input source **ri**  
**ri**      =      232 or 485

420

**CSC Clear all speed dial entries.**

This command is used to clear all speed dial entries. See the CDS, SD, CSE, CSF and CSN commands.

**CSC** clear all speed dial entries

**CSD Speed dial a number**

This command is used to speed dial a number.

See the CDS, CSC, CSE, CSF and CSN commands.

**CSD sn ?** prints the description for speed dial number sn

**CSD sn** speed dials speed dial number sn

sn = 3 digit speed dial number

**CSE Enter a number in the speed dial directory**

This command is used to insert a speed dial number in the directory

Not all combinations of entries are possible. The first decision is to determine the line format. This breaks down into two general categories and these are H221 and not H221.

For the H221 case, then the

- bit rate (**br**) determines the number of connected lines
- sampling rate (**sr**) must be 32, 44 or 48
- encoder algorithm (**ea**) must be MPEG2
- decoder algorithm (**da**) must be MPEG2

The actual bit rate used will be determined by the number of lines connected. The lines called must utilize 64 kbs only. H.221 cannot currently handle  $n * 56$  kbs.

For the L1 .. L6 case, then any of the rest of the parameters may be used. In this case, one phone number must be supplied. If more than one phone number is supplied, the data is broadcast to each of the connected lines.

For CCSL12 .. CCSL56, then

- bit rate (**br**) must be set to 112 or 128
- sampling rate (**sr**) must be 32 or 48
- encoder algorithm (**ea**) must be MPEG2, CCSO, CCSN

421

- decoder algorithm (da) must be MPEGL2, CCSO, CCSN
- two phone numbers (d1 and d2) must be supplied

See the CDS, CSC, CSD, CSF and CSN commands.

**CSZ na ?** print speed dial entry na

**CSZ na br sr ea em el NO d1 d2 d3 d4 d5 d6**

**CSZ na br sr ea em el YES da dl li d1 d2 d3 d4 d5 d6**

<b>na</b>	=	name of entry
<b>br</b>	=	24, 32, 40, 48, 56, 64, 80, 96, 112, 128, 144, 160, 192, 224, 256, 320, 384, A
<b>sr</b>	=	16, 22, 24, 32, 44 or 48
<b>ea</b>	=	da = MPEGL2, CCSO, CCSN or G.722
<b>em</b>	=	M, DM, JS, S
<b>el</b>	=	dl = COMH221, L1, L2, L3, L4, L5, L6, CCSL12, CCSL13, CCSL14, CCSL15, CCSL16, CCSL23, CCSL24, CSL25, CCSL26, CCSL34, CCSL35, CCSL36, CCSL45, CCSL46, CCSL56
<b>in</b>	=	YES or NO
<b>d1</b>	=	20 digit phone number
<b>d2</b>	=	20 digit phone number
<b>d3</b>	=	20 digit phone number
<b>d4</b>	=	20 digit phone number
<b>d5</b>	=	20 digit phone number
<b>d6</b>	=	20 digit phone number

**CSF Print first of speed dial entry**

422



description. If the optional parameter sh is set to A (abbreviated), only the entry number and speed dial description are displayed.

To print subsequent entries, see the CSN command.

This command prints the first entry in the speed dial list. The list is alphabetical by See the CDS, CSC, CSD, CSE and CSN commands.

**CSF [sh]** print first speed dial entry

sh = A

### CSI Set SPID for a Terminal Adaptor

This command is used to set the SPID for the digital interface. The SPID is only used for TA's in North America.

See the CAA, CAD, CCR, CCS, CDI, CHU, CLD, CTC and CTO commands

**CSI di ?** prints the SPID for digital interface di

**CSI di sd** set SPID for digital interface di to sd

di = 1, 2, ... 6

sd = 20 digit number

### CSL Set system loopback

This command is used to set the system into loopback. Individual digital interfaces may be looped back (see the CLB command ) but this command generates a loopback deeper inside the CODEC.

If the cdqPRIMA is powered down and powered up, the state of the loop back is NOT forgotten and the unit is set to to the state before the power was removed.

See the CBR and CLB commands.

**CSL ?** print system loopback state

**CSL sl** set system loop back state to state sl

sl = NORM or LB

### CSN Print next speed dial entry

This command prints the next entry in the speed dial list. The list is alphabetical by description. If the optional parameter sh is set to A (abbreviated), only the entry number and speed dial description are displayed.

See the CSF command for a description of the command output.

423



When the end of the list is reached, the message

**END OF LIST**

is displayed.

If the CSN command is given again after the END OF LIST is displayed, the first entry will be displayed. This means that continually entering the CSN command will repeatedly traverse the speed dial list.

See the CDS, CSC, CSD, CSE, and CSF commands.

**CSN [sh]**      Print next speed dial entry

sh      =      A

### **CST Report CODEC status**

This command reports the general status of the CODEC.

**CST**    report status

### **CSW Set switch type for a Terminal Adaptor**

This command is used to set the switch type for the TA

type of digital interface. The switch type refers to the telephone company central office switch. Switches are made by such companies such as AT&T, Northern Telecom, Siemens ... These switches run different versions of ISDN software. This command sets the TA to work with a particular type of switch software.

The digital interface, di, can be either of the interfaces for the port. For example, if the TA is in the DIF23 slot, then di can be either 2 or three to set the switch type.

See the CAA, CAD, CCR, CCS, CDI, CHU, CLD, CTC and CTO commands

**CSW di ?**      prints the switch type for digital interface di

**CSW di si**      set switch type for digital interface di to si

di              =      1, 2, ... 6

North America

si              =      NI1, 5E6, 5E8 or NTI

### **CTC Connect to TA control port**

424

9-22

BAD ORIGINAL

This command is used to connect to the TA control port. This allows access to all the TA functionality. In particular, it allows configuration and call monitoring.

The TA control ports are named as follows

DIF12	DIF1 and DIF2
tc	digital interface
DIF34	DIF3 and DIF4
DIF56	DIF5 and DIF6

If tc is set to DIF1, then communication is established via DIF1 to a far end cdqPRIMA. The far end cdqPRIMA must be in the ISDN communication mode. This can be done at the far end by executing the MFC command or by sending the in-band MFC command to the far end (>>MFC)

See the CAA, CAD, CCR, CCS, CDI, CHU, CLD, CSI and CTO commands

**CTC ?** print current TA connection

**CTC tc** set connection to TA tc

tc = NONE, DIF12, DIF34 or DIF56

### CTE Set TA remote control command response echo

This command is used to set the Terminal Adaptor (TA) remote control command echo. When downloading new software in flash, it is advisable to turn off command echo to speed the download process.

See the CTP commands.

**CTE ?** print TA remote control command response echo state

**CTE re** set TA remote control command response echo state to re

re = YES or NO

### CTI Set Time Code readout source

This command is used to display the Time Code on the LCD display.

The displayed Time Code can be the Time Code input to the encoder, or the Time Code output from the decoder or no time code displayed.

425

9-23

If the timecode is displayed on the display, then depressing any front panel key or issuing the CTI NONE command terminates the Time Code display on the LCD.

This command is useful to check if time code is being received correctly by the encoder or the decoder.

See the CTL, CTS and CTT commands.

**CTI ?**            print Time Code readout source  
**CTI ti**           set Time Code readout source to ti  
                  ti        =        NONE, INPUT, OUTPUT

#### **CTL Print last Time Code received**

This command is used to display the last time code received.

See the CTI, CTS and CTT commands.

**CTL tf** print last time code received for source tf  
                  tf        =        INPUT or OUTPUT

#### **CTM set timer timeout duration**

This command is used to set an internal timer. This timer is used by the PRIMA Logic Language (PLL) to generate events.

This command starts the specified timer tn for the duration ti

The duration is in seconds.

See the CCT and CEA commands.

**CTM tn ?**        print timer tn time left in seconds  
**CTM tn tl**       set timer tn to timeout in tl seconds  
                  tn        =        0 or 1  
                  tl        =        0..999999

#### **CTO Set TA dialing timeout**

This command is used to set the terminal adaptor dialing timeout. This timeout is used to terminate the dialing sequence for an individual TA.

See the CAA, CAD, CCR, CCS, CDI, CHU, CLD, CSI and CTC commands

**CTO ?**            print TA dialing timeout value

426

9-24

BAD ORIGINAL



**CTO to** set TA dialing timeout value to (in seconds)  
**to =** 5..24

### CTP Set TA remote control protocol usage

This command forces the control processor to use protocol protected communication on all TA remote control messages. If no protocol is used, then point to point communications is assumed (a pc is connected to only 1 encoder). If protocol is used, then each CODEC device must have an id set by the CID command. The protocol can then select the specified device. Protocol communication can be used for point to point and point to multipoint communication.

If protocol was not enabled and it is enabled, the response will be in protocol mode (even though the input command was not in protocol mode) with a BSN of 0.

See the CTE commands.

**CTP ?** print TA remote control protocol mode  
**CTP tp** set TA remote control protocol mode to tp  
**tp =** YES or NO

### CTS Print Time Code speed

This command is used to display the Time Code speed. Time code can be 24, 25 or 30 frames per second.

See the CTI, CTL and CTT commands.

**CTS tf** print the time code speed for source tf  
**tf =** INPUT or OUTPUT

### CTT Enable/disable Time Code

This command is used to enable or disable the time code feature.

In the US, time code frames are transmitted at 30 frames per second. In Europe, they are transmitted at 24 frames per second.

The time code sub-system in the cdqPRIMA automatically senses and adapts to the input time code rate.

If time code is present at the encoder, the PRIMA attempts to deliver it to the far end decoder. To do this the ancillary data channel is used. This requires approximately 2400 bits per second of ancillary data.

427



If **tt** is set to OFF, then the time code input is always ignored and no ancillary data channel capacity is used.

If **tt** is set to ON and there is no time code signal present at the input, then no ancillary data resources are utilized.

See the CTI, CTL and CTS command.

**CTT ?** print current Time Code type

**CTT tt** set time code type to **tt**

**tt** = ON or OFF

### CVA Define virtual action

This command is used to set a virtual action. The command associated with the virtual action is executed when the associated action becomes true. The Event - Action logic determines when an action becomes true.

For example

CVA 2 CSD 5

assigns the PRIMA Remote Control Command

CSD 5

to virtual action 2.

**CVA va ?** print command associated with virtual action **va**

**CVA va cm** define virtual action **va** as the command **cm**

**va** = 0 . . 3

**cm** = any PRIMA Remote Control Command (PRCC)

### CVN Print software version number

This command is used to print the software version number for a thing in FLASH ram.

**CVN tx** print version and verify checksum of thing **tx**

**tx** = DSPD, DSPDX, DSPDXX, DSPV,  
DSPE, DSPEX, DSPR, CP or CPX

### CVU Set level meter mode

428

This command selects the level meter mode. The level meter can be used to display the level of the audio as a normal vu meter. It can also be used to display the magnitude of the input as well as the position of the stereo image.

**CVU ?**            print current level meter mode.  
**CVU vu**           set level meter mode to vu  
                     **vu**       =       LEVEL, IMAGE or PHASE

#### **DAL Set decoder algorithm**

This command is used to set decoder algorithm.

See the DBR, DCO, DCS, DDA, DDO, DIN, DLI, DMD, DMU and DSP commands.

**DAL ?**            print decoder algorithm  
**DAL al**           set decoder algorithm to al  
                     **al**       =       MPEGL2, CCSO, CCSN or G.722

#### **DBR Set decoder bit rate**

This command is used to set the decoder digital audio bit rate.

Normally, the decoded bit stream dictates the sampling rate when in the decoder operates independently from the encoder (see the DIN command). This command has no effect when DIN is set to NO.

Setting br to A lets the cdqPRIMA choose the bit rate based on the clock and the line type (see the DLI command).

See the DAL, DCO, DCS, DDA, DDO, DIN, DLI, DMD, DMU and DSP commands.

**DBR ?**            print decoder bit rate  
**DBR br**           set decoder bit rate to br  
                     **br**       =       24, 32, 40, 48, 56, 64, 80,  
    96, 112, 128, 144, 160, 192,  
    224, 256, 320, 384, A

#### **DCO Set decoder decoding mode**

This command is used to control decoding of audio bit streams.

If ISO is selected, then only ISO layer 2 bit streams are decoded.

If ISOCCS is selected, then ISO layer 2 and older CCS bit streams are decoder. This command is for compatibility checking of bit streams.

See the DAL, DBR, DCS, DDA, DDO, DIN, DLI, DMD, DMU, DRS and DSP commands.

**DCO ?**            print decoder decoding mode  
**DCO co**           set decoder decoding mode  
**co**            =        ISO or ISOCCS

### **DCS Set channel copy/swap mode**

This command is used to control the audio output. It allows the left channel to be copied over the right channel (CLTOR), the right channel to overwrite the left channel (CRTOL) or the left and right channels to

be swapped (SWAP). If cs is set to NONE, then the output of the decoder is the same as received, ie. left channel to left channel and right channel to right channel.

This command is useful for controlling the action of the cdqPRIMA in the presence of mono audio signals.

See the DAL, DBR, DCO, DDA, DDO, DIN, DLI, DMD, DMU and DSP commands.

**DCS ?**            print decoder copy/swap mode  
**DCS cs**           set decoder copy/swap mode to cs  
**cs**            =        NONE, CLTOR, CRTOL, SWAP

### **DDA Calibrate DA converter**

This command is used to calibrate the DA converter. This operation takes about .1 second and during the calibration process, the audio output is muted. The DA converter is calibrated during power up but can be recalibrated at any time.

See the DAL, DBR, DCO, DCS, DDO, DIN, DLI, DMD, DMU and DSP commands.

**DDA**            calibrate da converter

### **DDO Set digital output sampling rate.**

430



The digital audio is output from the MUSICAM decoder at the sampling rate specified in the MUSICAM header. This rate can then be converted to other rates via a sample rate converter.

The sample rate converter is capable of sampling rate changes between .51 and 1.99. For example, if the MUSICAM receiver received a bit stream which indicated that the sampling rate was 24 kHz, then the output sampling rate could be set to 32 or 44 kHz but not 48 kHz since 48 kHz would be a sampling rate conversion of 2.0 to 1. This is out of the range of the sampling rate converter.

The following table outlines the valid sampling rate conversions.

Input Sampling Rates	Output Sampling Rates			
	29.5	32	44.1	48
16	X			
22.05	X	X		
24	X	X	X	
32	X	X	X	X
44.1	X	X	X	X
48	X	X	X	X

Notice that the 16 kHz sampling rate cannot be output via the AES/EBU output port since it cannot be sample rate converted to any allowed value.

This command sets the digital audio (AES/EBU, SPDIF or optical) sampling rate. Setting do to M means that the sampling rate should follow the value contained in the MUSICAM audio frame.

See the DAL, DBR, DCO, DCS, DDA, DIN, DLI, DMD, DMU and DSP commands.

DDO ?      print the decoder output sampling rate  
 DDO do     set digital output sampling rate to do  
 do        =    29, 32, 44 or 48

DES Enable decoder AES sync timing

431

This command is used to enable/disable the use of the decoder AES/EBU sync signal. Normally, the AES/EBU sync signal for the decoder is used to determine the rate of the output of the AES/EBU decoder output. The AES/EBU decoder sync input can be ignored by setting es to DISABLE. See Fig. 7a, 7b and 7c for reference.

If there is no cable connected to the decoder sync input or DES is set to DISABLE, then the DA converter and the AES/EBU transmitter in the decoder is timed off the network clock. The exact value of the clock is phase locked to the network clock at a rate given by information in the received ISO/MPEG data stream.

If there is a sync signal present at the decoder sync input, then the signal going to the decoder DA converter and to the AES/EBU transmitter is rate adapted to the frequency of the the received sync input.

DES ?            print status of decoder AES sync timing  
 DES es           enable decoder AES sync timing  
                  es        =        REQ or NOTREQ

#### DHV Set decoder headphone volumn level

This command is used to set the decoder volumn level. The level applies when the decoder is selected as the source of audio output to the headphone jack.

DHV ?            print decoder headphone volumn level  
 DHV hv           set decoder headphone volumn to hv  
                  hv        =        0 .. 127, + or -

#### DIN Set decoder - encoder interaction

This command is used to control the interaction between the decoder and the encoder. If in is set to NO, then the decoder and encoder interact. This is necessary for H.221 and one mode of two line CCS inverse multiplexing.

If ELI is set to COMH221, DIN is automatically set to NO.

Setting in to YES forces the decoder to operate completely indepently from the encoder. Any operation of the encoder has no effect on the decoder and visa versa.

See the DAL, DBR, DCO, DCS, DDA, DDO, DLI, DMD, DMU and DSP commands.

DIN ?            print decoder - encoder interaction

432

9-30

BAD ORIGINAL



**DIN in**      set decoder - encoder interaction to in  
                  in      =      YES or NO

### **DLI Set decoder digital lines format**

This command sets the format for the decoder digital interface lines.

This command is only valid if the decoder is set to operate independently. See the DIN command.

The ls parameter is defined as follows:

L1 indicates that only line 1 should be used.

L2 indicates that only line 2 should be used.

L6 indicates that only line 6 should be used.

CCSL12 .. CCSL56 indicates that CCS two line combined mode is to be used (see the ELI command).

See the DAL, DBR, DCO, DCS, DDA, DDO, DIN, DMD, DMU and DSP commands.

**DLI ?**      print current decoder digital line format.

**DLI ls**      set decoder digital line format ls

ls      =      L1, L2, L3, L4, L5, L6,  
                  CCSL12, CCSL13, CCSL14,  
                  CCSL15, CCSL16,  
                  CCSL23, CCSL24, CCSL25,  
                  CCSL26, CCSL34,  
                  CCSL35, CCSL36, CCSL45,  
                  CCSL46, CCSL56

### **DMD Set decoder maintenance diagnostic mode**

This command is used to generate an output tone from the decoder. This is useful for setting levels in an analog system. It is also useful for checking the DA and digital outputs.

A 1000 and a 9600 Hz tone can be output to the left, right or both channels.

**433**

See the DAL, DBR, DCO, DCS, DDA, DDO, DIN, DLI, DMU and DSP commands.

If **md** is set to NORM, then the normal audio is output.

**DMD ?** print decoder maintenance diagnostic mode

**DMD md** set decoder maintenance diagnostic mode to **md**

**md** = NORM, 1KLEFT, 1KRIGHT, 1KBOTH,  
10KLEFT, 10KRIGHT, 10KBOTH

### DMU Mute decoder output channels

This command is used to mute the decoder audio output channels.

See the DAL, DBR, DCO, DCS, DDA, DDO, DIN, DLI, DMD and DSP commands.

**DMU ?** print the channels muted

**DMU mu** mute decoder outputs **mu**

**mu** = LEFT, RIGHT, BOTH or NONE

### DQQ Print command summary for decoder commands

This command is used to print a summary of all the Dxx commands. See the CQQ, EQQ, MQQ and QQQ (HELP) commands.

**DQQ** print command summary

### DRS Print decoder real-time status bits

This command is used to print the decoder status bits from the ISO/MPEG frame header. The emphasis, copyright, private, protection and copy bits are displayed by this command.

If the decoder is not framed, then the words

NOT FRAMED

are displayed.

If the decoder is framed, then the following is displayed.

ee o w v mm

The ee characters are one of the following

ee	description
----	-------------

434



NONE	no emphasis
50/15	50/15 microsecond emphasis
RES	reserved
J.17	CCITT J.17 emphasis

The o character is one of the following

<i>o</i>	<i>description</i>
O	original version
C	copyed version

The w character is one of the following

<i>w</i>	<i>description</i>
W	copyrighted version
	non-copyrighted version

The v character is one of the following

<i>v</i>	<i>description</i>
V	the private bit is on
	the private bit is off

The mm characters are one of the following

<i>mm</i>	<i>description</i>
PC	CRC algorithm is old ISO and frame type is CCS
PM	CRC is th old ISO and the frame type is ISO
MC	CRC is ISO and the frame type is CCS

435

MM	CRC is ISO and frame type is ISO
NC	there is no crc on the frame

See the DBR, DCO, DCS, DDA, DDO, DIN, DLI, DMD, DMU and DSP commands.

**DRS ?** print decoder real-time status bits

**DRS** print decoder real-time status bits

### **DSB Set decoder synchronous ancillary data bit rate**

This command is used to set the decoder synchronous ancillary data bit rate. If the decoder is not independent, then the decoder synchronous ancillary data bit rate is set by the ESB command. If the decoder is independent, then the decoder synchronous ancillary data bit rate is set by this command.

See the CAN, CDR, DIN and ESB commands.

**DSB ?** print decoder synchronous ancillary data bit rate

**DSB sb** set decoder synchronous ancillary data bit rate to sb

sb = 8, 16, 32 or 64

### **DSP Scale factor protection**

This command is used to enable or disable the use of scale factor protection. If scale factor protection checking is disabled, abit errors can have a much greater effect on the audio output than if scale factor protection is used.

If scale factor protection is used by the decoder, the encoder must also have scale factor protection enabled.

See the DAL, DBR, DCO, DCS, DDA, DDO, DIN, DLI, DMD and DMU commands.

**DSP ?** print decoder scale factor protection status

**DSP sp** set decoder scale factor protection to sp

sp = YES or NO

### **DTI Decoder timing**

This command sets decoder timing source. See the Timing Section for a detailed description of this command.

436

**DTI ?**            print decoder timing source  
**DTI ts**           set decoder timing source ts  
                  ts       =       NORMAUTO, INTAUTO, INT or AES

### **EAD Calibrate AD converter**

This command is used to calibrate the AD converter. This operation takes about .1 second and during the calibration process, the audio output is muted.

The A-D converter is calibrated at power up. Calibrating the A-D converter before a critical recording results in the highest possible quality.

See the EAB, EAI, EAL, EAM, EBR, ELI, ESP and ESR commands.

**EAD**    calibrate ad converter

### **EAI Set encoder audio input source**

This command selects the type of input to the encoder. It can be either an analog or a digital input. The type of digital input (AES/EBU, SPDIF or optical) is selected by switches on the encoder.

See the EAB, EAD, EAL, EAM, EBR, ELI, ESP and ESR commands.

**EAI ?**            print current encoder audio source  
**EAI ai**           set encoder audio source ai  
                  ai       =       A or D

### **EAL Set encoder algorithm**

This command is used to set encoder algorithm. MPEG L2 set the encoder to output ISO/MPEG layer 2 frames. CCSN outputs CCS "new" frames. CCSO outputs CCS "old" frames. G.722 outputs the G.722 algorithm.

The various CCS algorithms are variations of the ISO/MPEG layer 2 standard. They were implemented before the standard was finalized and are included for backward compatibility with older CDQ200x CODEC's.

See the EAB, EAD, EAI, EAM, EBR, ELI, ESP and ESR commands.

**EAL ?**            print encoder algorithm  
**EAL al**           set encoder algorithm to al  
                  al       =       MPEG L2, CCSO, CCSN or G.722

437

**EAM Set encoder algorithm mode**

This command is used to set encoder algorithm mode when the algorithm is MPEGL2, CCSO or CCSN. See the EAL command.

See the EAB, EAD, EAI, EAL, EBR, ELI, ESP and ESR commands.

**EAM ?** print encoder algorithm mode

**EAM am** set encoder algorithm mode to **am**

**am** = S (stereo), JS (joint stereo),  
DM (dual mono) or M (mono)

**EBR Set encoder bit rate**

This command is used to set the encoder digital audio bit rate.

If A is selected, the bitrate is determined by the input digital interface clock and the line format (ELI).

If ELI is set to COMH221, the EBR is automatically to the necessary bitrate to match the number of connected lines. The bit rate set by this command is ignored in the COMH221 mode.

Upon changing to any line format, the bitrate will be set to the bitrate set by this command.

See the EAB, EAD, EAI, EAL, EAM, ELI, ESP and ESR commands.

**EBR ?** print encoder bit rate

**EBR br** set encoder bit rate to **br**

**br** = 24, 32, 40, 48, 56, 64, 80,  
96, 112, 128, 144, 160, 192,  
224, 256, 320, 384, A

**ECR Set encoder copyright bit in header**

This command is used to enable or disable copyright bit in the ISOheader.

See the EEP, EOR and EPR commands.

**ECR ?** print encoder copyright bit status

**ECR cr** set encoder copyright bit status to **cr**

**cr** = YES or NO

**EEP Set encoder emphasis bit in header**

438

This command is used to enable or disable emphasis bit in the ISO header.

See the ECR, EOR and EPR commands.

**EEP ?**            print encoder emphasis bit status  
**EEP ep**           set encoder emphasis bit status to ep  
                   ep     =     NO, 50, or J.17

#### **EHV Set encoder headphone volume level**

This command is used to set the encoder volume level. The level applies when the encoder is selected as the source of audio output to the headphone jack.

See the CHP, CHV and DHV commands.

**EHV ?**            print encoder headphone volume level  
**EHV hv**           set encoder headphone volume to hv  
                   hv     =     0 .. 127, + or -

#### **ELI Set encoder digital lines format**

This command sets the format for the encoder digital interface lines.

The li parameter is defined as follows:

COMH221 indicates that multiple lines are combined utilizing H.221 L1 indicates that only line 1 should be used.

L2 indicates that only line 2 should be used.

L6 indicates that only line 6 should be used.

CCSL12 .. CCSL56 indicates that CCS two line combined mode is to be used.

If COMH221 is selected, usage of a maximum 6 lines is possible.

The actual number of lines is determined by the number of lines dialed. The encoder/decoder bit rate is set to 384 kbs and the decoder is set to not independent (DIN NO).

The TA lines are set **CONNECTED** when that are manually dialed, automatically dialed or connected to an incoming call.

See the EBR and DIN commands.

**ELI ?** print current encoder digital line format.

**ELI li** set encoder digital line format (li)

li = COMH221, L1, L2, L3, L4, L5, L6,  
CCSL12, CCSL13, CCSL14, CCSL15,  
CCSL16, CCSL23, CCSL24, CCSL25,  
CCSL26, CCSL34, CCSL35, CCSL36,  
CCSL45, CCSL46, CCSL56

### ELU Set link message update rate

This command is used to set the link message update rate. The link messages are the exported part of the action word.

Link messages are sent to the far end cdqPRIMA every time the action word changes. If no changes in the action occur, the the link message is sent at a rate given by ru.

Ru of 1 means link message updates every .1 second, while ru = 5 means update link messages every .5 second.

An update rate of 0 turns off link messages.

See the CEV, CEA, CAR, CLA, CRA and ESW commands.

**ELU ?** print link message update rate

**ELU ru** set the link message update rate to ru

ru = 0 .. 10

### EOR Set encoder original bit in header

This command is used to enable or disable original bit in the ISO header.

See the ECR, EEP and EPR commands.

**EOR ?** print encoder original bit status

**EOR or** set encoder original bit status to or

or = YES or NO

### EPB Load all default psychoacoustic parameters

440

This command is used to load the default (factory supplied) psychoacoustic parameters. This is done by setting the tables for each sampling rate and bit rate to point to the factory supplied parameters.

This command is the same as executing the following two commands for each possible sampling rate and bit rate.

**EPD br sr**

to find the default table number for the sampling rate **sr** and bit rate **br**

**EPT tb br sr**

to set table **tb** (found from the above command) as the table to be used for sampling rate **sr** and bit rate **br**

See the EPD, EPL, EPP, EPS and EPT commands.

**EPB** load all default psychoacoustic parameters

#### **EPD Get default psychoacoustic parameter table number**

This command is used to get the default psychoacoustic parameter table number for the specified bit and sampling rates. The table number will be between 120 and 239. It also returns a second number to the right of the first number. This number is the suggested table number for the user defined bit rate and sampling rate. This suggested table number can be ignored.

See the EPL, EPP, EPS, EPT and EPY commands.

**EPD br sr** get default psychoacoustic parameter table number

**br** = 24, 32, 40, 48, 56, 64, 80,  
96, 112, 128, 144, 160, 192,  
224, 256, 320, 384

**sr** = 16, 22, 24, 32, 44 or 48

#### **EPI Set encoder private bit in header**

This command is used to enable or disable private bit in the ISO header.

See the ECR, EEP, EOR and EPI commands.

**EPI ?** print encoder private bit value

**EPI pb** set encoder private bit value to **pb**

**pb** = ON or OFF

#### **EPL Load psychoacoustic parameter table from flash**

441

This command is used to load psychoacoustic parameters from FLASH into RAM memory. These parameters become the current parameters and are downloaded to the encoder.

See the EPD, EPP, EPS, EPT and EPY commands.

```
EPL tb      load psychoacoustic parameter table from flash table tb
tb      =      0.239
```

### EPP Set psychoacoustic parameter

This command is used to set a pshchoacoustic parameter.

The parameter type (EPY) must be set for each parameter before this command can be used.

See the EPD, EPL, EPS, EPT and EPY commands.

**ESP pp ?**                      print the value of psychoacoustic parameter pp

**ESP pp pv [0]** set psychoacoustic parameter pp to value pv with optional type 0 indicating pv is in hex

pp = 0.31

**pv** = floating point or integer number .

### EPR Set encoder protection bit in header

**This command is used to enable or disable protection bit in the ISO header.**

See the ECR, EEP, EOR and EPR commands.

**EPR ?**      print encoder protection bit status

**EPR pr** set encoder protection bit status to pr

**pr** = YES or NO

### EPS Store psychoacoustic parameter table in flash

This command is used to store the current psychoacoustic parameters into flash memory.

Table numbers from 0 to 119 are the normal user tables. Table numbers from 120 to 239 are the default psychoacoustic tables and can only be overwritten by the system administrator.

See the EPD, EPL, EPP, EPT and EPY commands.

**EPS tb** store psychoacoustic parameter table into flash table tb



**tb** = 0..119 for normal users

**tb** = 0..239 for system administrators

### **EPT Assign psychoacoustic parameter table**

This command is used to assign a psychoacoustic parameter table to be used for a specified bit rate and sampling rate.

Psychoacoustic parameter tables are numbered from 0 to 239. Tables 0..119 hold user defined tables while tables 120..239 hold the system default tables.

If the **EPT tb ?** command is entered, pairs of numbers are returned. The left hand number is the bit rate and may be any value specified by

the **br** field. The right hand number is the sampling rate and may be any of the values specified in the **sr** field.

If no numbers are returned, then the specified table is not used by any sampling and bit rate.

If multiple pairs of numbers are returned, then the specified psychoacoustic table is used by more than one sampling / bit rate.

See the **EPD**, **EPL**, **EPP**, **EPS** and **EPY** commands.

**EPT tb ?** print the bit rate and sampling rate for table **tb**

**EPT tb br sr** assign psychoacoustic parameters table **tb** to be used for sampling rate **sr** and bit rate **br**

**tb** = 0 .. 239

**br** = 24, 32, 40, 48, 56, 64, 80,  
96, 112, 128, 144, 160, 192,  
224, 256, 320, 384

**sr** = 16, 22, 24, 32, 44 or 48

### **EPY Set psychoacoustic parameter type**

This command is used to set the psychoacoustic parameter type. This command is used in conjunction with the **EPP** command.

See the **EPD**, **EPL**, **EPP**, **EPS** and **EPT** commands.

**EPY pp ?** print psychoacoustic parameter type

**EPY pp pt** set psychoacoustic parameter **pp** to type **py**

**pp** = 0..31

443

pt = 0..4

### **EQQ Print command summary for encoder commands**

This command is used to print a summary of all the Exx commands.

See the CQQ, DQQ, MQQ and QQQ (HELP) commands.

**EQQ** print command summary

### **ESB Set encoder synchronous ancillary data bit rate**

This command is used to set the encoder synchronous ancillary data bit rate. If the decoder is not independent, then the decoder synchronous ancillary data bit rate is also set to the same value.

See the CAN, CDR, DIN and DSB commands.

**ESB ?** print encoder synchronous ancillary data bit rate

**ESB sb** set encoder synchronous ancillary data bit rate to sb

sb = 8, 16, 32 or 64

### **ESP Set encoder scale factor protection**

This command is used to enable or disable the use of scale factor protection. If scale factor protection checking is disabled, a bit errors can have a much greater effect on the audio output than if scale factor protection is used.

If scale factor protection is used by the decoder, the encoder must also have scale factor protection enabled. Scale factor protection can be enabled in the encoder and not enabled by the decoder.

See the EAB, EAD, EAI, EAL, EAM, EBR, ELI and ESR commands.

**ESP ?** print encoder scale factor protection status

**ESP sp** set decoder scale factor protection to sp

sp = YES or NO

### **ESR Encoder sampling rate**

This command sets the sampling rate for the A-D converter or the digital audio input.

This only applies for the MPEG2, CCSN and CCSO algorithms. For G.722 the sampling rate is fixed at 16 kHz.

See the EAB, EAD, EAI, EAL, EAM, EBR, ELI, ESP and ESR commands.

444

**ESR ?** print current encoder sampling rate

**ESR sr** set encoder sampling rate (sr) to one of the following:

sr = 16, 22, 24, 32, 44 or 48

### ESW Set a simulated switch

This command is used to simulate a contact closure. This command causes actions based on the Event-Action logic.

See the CEV, CEA, CAR, CLA, CRA, and ELU commands.

**ESW sw ?** print status of simulated switch number sw

**ESW sw ss** set simulated switch number sw to state ss

sw = CI0 .. CI7

ss = ON or OFF

### ETI Encoder timing

This command sets encoder timing source. The three choices are normal, internal crystal clock and aes/ebu.

**ETI ?** print encoder timing source

**ETI te** set encoder timing source ts

te = NORM, INT or AES

### MBC Display BER counter

This command displays the BER counter. See the MBD, MBL, MBR and MBU commands. **MBC ?** Display the BER counter

**MBC** Display the BER counter

### MBD Set BER down count rate

This command is used to set the BER down count rate. It is used in conjunction with the MBL command. For a detailed explanation of the MBD command, see the MBL command.

See the CEA, MBC, MBU, MBR and MBL commands.

**MBD ?** print current BER down count rate

**MBD bd** set BER down count rate to bd

bd = 0 .. 9

445



## MBL Set BER count rate limit

This command is used to set the threshold limit for bit error rate.

If the bit error rate counter goes above this limit, then the BER event is set to true.

Each time a decoded frame is received (every 24 ms for 48k sampling rate MPEG I), the status of the BER bit is checked. The BER bit is set to a 1 by the decoder if MPEG frame protection is found and the frame CRC is in error. If the BER bit is on, the BER counter is incremented by the value set by the MBU command. If the BER bit is off, then the BER counter is decremented by the value set by the MBD command. When the BER counter is equal or above the level set by the MBL command, then the BER event is set to true, otherwise it is set to false.

The contents of the BER counter may be displayed by the MBC command.

The BER counter may be set to 0 by the MBR command.

In a typical application of the BER commands, the following commands are used

MBU 1	set to count up by one on each frame with an error
MBD 0	set not to count down on ok frames
MBR	clear the counter
MBL 1234	wait until the ber count goes to 1234

The above sequence of commands can be used count the total number of bit errors and set the BER event when the count goes above 1234. The above sequence has the drawback that it never resets the BER count in the presence of good frames. The following remedies the situation by providing a leaky counter.

MBU 10	set to count up by one on each frame with an error
MBD 1	set not to count down on ok frames
MBR	clear the counter
MBL 12340	wait until the ber count goes to 12340

In the case above, every time a frame with a BER occurs, the count increments by 10. If a good frame occurs, then the count decrements by one. A long string of good frames erases a bad frame.

See the CEA, MBC, MBD, MBR and MBU commands.

MBL ?	print current BER up count rate
MBL bl	set BER up count rate to bl

446

bl = 0 .. 32767

### MBM Boot the cdqPRIMA from ROM

This command is can only be used when the cdqPRIMA is executing out of the FLASH. It is used to boot the cdqPRIMA so that it runs out of ROM. In the ROM mode, all software can be downloaded including the control processor.

This command can be used to force the control processor into the software download mode. When control is passed to the ROM boot, the rear panel remote control port (RC) is connected to either the usual rear panel connector (RP) or digital interface port 1 (DIF1).

**MBM rp** boot the cdqPRIMA out of ROM and set RC port to rp  
 rp = RP or DIF1

### MBO Boot the cdqPRIMA

This command is can only be used when the cdqPRIMA is executing out of the ROM. It is used to boot the cdqPRIMA so that it runs out of FLASH and has full functionality.

This command can be used after downloading new software into FLASH.

**MBO** boot the cdqPRIMA out of FLASH

### MBR Reset BER counter

This command set the BER counter to 0.

See the CEA, MBC, MBD, MBL and MBU commands.

**MBR** Reset BER counter

### MBU Set BER up count rate

This command is used to set the BER up count rate. It is used in conjunction with the MBL command. For a detailed explanation of the MBU command, see the MBL command.

See the CEA, MBC, MBD, MBR and MBL commands.

**MBU ?** print current BER up count rate

**MBU bu** set BER up count rate to bu

bu = 0 .. 9

### MCP Set connect port

447

This command is used to connect the current remote port to a connect port. This allows a direct RS232 connection from the remote port the to connect port. This allows manual control of the connected port.

The remote port is the front panel or the rear panel remote control port.

**MCP ?** print current connect port

**MCP cp** set connect port to cp

cp = NONE, 0 ... 7, TA0, TA1 or TA2

### **MET Enable hardware tests**

This command is used to enable hardware tests. When hardware tests are enabled, then the normal operation of the the cdqPRIMA hardware is disabled. If hardware tests are enabled, then the various hardware subsystems may be tested.

See the MTM command.

**MET et** enable hardware tests

et = ENABLE or DISABLE

### **MHT Perform hardware tests**

This command is used to perform hardware tests.

Setting ht to ALL performs all hardware tests See the MET and MTM command.

**MHT ht** perform hardware test ht

ht = ALL, TC, IO, LED

### **MOC Display OOF counter**

This command displays the OOF counter.

See the MOD, MOL, MOR and MOU commands. **MOC ?** Display the OOF counter

**MOC** Display the OOF counter

### **MOD Set OOF down count rate**

This command is used to set the OOF down count rate. It is used in conjunction with the MBL command. For a detailed explanation of the MOD command, see the MBL command.

See the MOC, MOU, MOR and MOL commands.

448

**MOD ?**            print current OOF down count rate

**MOD od**           set OOF down count rate to **od**

**od**        =        0 .. 9

### **MOL Set OOF count rate limit**

This command is used to set the threshold limit for bit error rate.

If the bit error rate counter goes above this limit, then the OOF event is set to true.

Each time a decoded frame is received (every 24 ms for 48k sampling rate MPEG I), the status of the OOF bit is checked. The OOF bit is set to a 1 by the decoder if MPEG frame protection is found and the frame CRC is in error. If the OOF bit is on, the the OOF counter is incremented by the value set by the MOU command. If the OOF bit is off, then the OOF counter is decremented by the value set by the MOD command. When the OOF counter is above the level set by the MOL command, then the OOF event is set to true, otherwise it is set to false.

The contents of the OOF counter may be displayed by the MOC command.

The OOF counter may be set to 0 by the MOR command.

In a typical application of the OOF commands, the following commands are used

**MOU 1**            set to count up by one on each frame with an error

**MOD 0**            set not to count down on ok frames

**MOR**             clear the counter

**MOL 1234**        wait until the ber count goes to 1234

The above sequence of commands can be used count the total number of bit errors and set the OOF event when the count goes above 1234.

The above sequence has the drawback that it never resets the OOF count in the presence of good frames. The following remedies the situation by providing a leaky counter.

**MOU 10**           set to count up by one on each frame with an error

**MOD 1**            set not to count down on ok frames

**MOR**             clear the counter

**MOL 12340**        wait until the ber count goes to 12340

449

In the case above, every time a frame with a OOF occurs, the count increments by 10. If a good frame occurs, then the count decrements by one. A long string of good frames erases a bad frame.

See the MOC, MOD, MOR and MOU commands.

**MOL ?** print current OOF up count rate  
**MOL ol** set OOF count rate limit to ol  
 ol = 0 .. 32767

### MOR Reset OOF counter

This command set the OOF counter to 0.

See the MOC, MOD, MOL and MOU commands.

**MOR** Reset OOF counter

### MOU Set OOF up count rate

This command is used to set the OOF up count rate. It is used in conjunction with the MOL command. For a detailed explanation of the MOU command, see the MOL command.

See the MOC, MOD, MOR and MOL commands.

**MOU ?** print current OOF up count rate  
**MOU ou** set OOF up count rate to ou  
 ou = 0 .. 9

### MPD Display peak detector level

This command is used to read the level of the peak detector.

The value of the peak is measured in dB down from the maximum. For example a peak reading of -10 indicates that the highest peak value since the last peak level status request was -10 dB.

The largest value the peak can be is 0 dB.

Once the the peak value is read, it is set to -150 dB.

**MPD pd** read peak detector level in dB down from maximum.  
 pd = EL, ER, DL or DR

### MQC Display quiet detector level time left

450

9-48

BAD ORIGINAL



This command is used to read the the quiet detector time left counter. This is the time left in seconds before the specified input is declared as quiet.

If the time returned is between 0 and 255. If it is 255, then the quiet time has been set to 0 and the quiet detector for the input has been disabled.

See the CEA, MQD, MQL, MQD commands.

**MQC qd** read quiet detector time left on input **qd**  
**qd** = EL, ER, DL, DL, E or D

### MQD Display quiet detector level

This command is used to read the level of the quiet detector. This allows the monitoring of the average level of the audio signal averaged over 1 second.

The value reported is in dB down from the maximum value. Thus a value of -12 dB represents -12 dB down from the highest value.

The largest value returned is 0 dB.

The quiet detector level readings are updated approximately every 1 second. This means that if the MQD command is issued more often than once per second, it will return the same value.

A qd value of E means encoder left or encoder right, whichever has the highest value. If the encoder left channel has a level has a quiet detector level of -87 dB and the encoder right channel has a value of -33 dB, the command MQD E would return a value of -33. A similar definition applies for the D command.

See the CEA, MQC, MQL and MQT commands.

**MQD qd** read quiet detector level in dB down from maximum.  
**qd** = EL, ER, DL, DR, E or D

### MQL Set quiet detector level

This command is used to set the threshold level for silence detection. The input audio level must be below this threshold for a certain period of time to be considered as a silent input.

The time duration is set by the MQT command.

See the CEA, MQD, MQT and MQC commands.

**MQL qd ?** print quiet level for input **qd**  
**MQL qd ql** set the quiet level in dB relative to maximum input to ql for input **qd**

451



**qd** = EL, ER, DL, DR, E or D

**ql** = -1 to -120

### **MQQ Print command summary for decoder commands**

This command is used to print a summary of all the Mxx commands.

See the CQQ, DQQ, EQQ and QQQ (HELP) commands.

**MQQ** print command summary

### **MQT Set quiet time duration**

This command is used to set the time in seconds that the input level must be below the threshold level before it is considered to be silent. The threshold level is set by the MQL command.

See the CEA, MQC, MQD and MQL commands.

**MQT qd ?** print quiet time duration for input qd

**MQT qd qt** set the quiet time duration to qt for input qd

**qd** = EL, ER, DL, DR, E or D

**qt** = 0 (to set no quiet level checking on input qd)  
1 . . . 254 (number of seconds of quiet)

### **MRM Boot the far end cdqPRIMA from ROM**

This command is used to force the far end cdqPRIMA to execute from boot ROM. This allows the far end to accept download information.

CAN must be set to mode 2 and the near and far end cdqPRIMA's must be operating in MPEGL2, CCSO or CCSN for proper operation.

See the CAN command.

**MRM** boot the far end cdqPRIMA out of ROM

### **MRS Set rear panel remote control uart source**

This command is used to set the source for the rear panel remote control UART. This UART may be connected to the rear panel connector or to DIF1. If the rear panel is selected, then the CRB command determines the baud rate for the port. If DIF1 is selected, then the clock on that DIF determines the bit rate. The later case is used for remote downloading of software via the DIF (ISDN).

See the MBM and MRM commands.

452



**CRS ?**            print rear panel remote control UART source

**CRS rp**           set rear panel remote control UART source to rp

rp        =        RP or DIF1

### **MSY Synchronize RAM and BBM**

This command is used to write any unwritten bytes to nonvolatile memory. Many of the variables that are kept in nonvolatile are first written to standard RAM and at a later time, they are flushed to battery backed up RAM (BBM). This command forces all bytes which are in ram but not in BBM to be written.

This command can be issued just before turning off the power to insure that all "dirty" bytes are written to RAM.

**MSY**    synchronize RAM and BBM.

### **MTM Perform a test measurement**

This command is used to perform a test measurement.

See the MET command.

**MTM ?**            print current test measurement in progress

**MTM tm**           start test tm

tm        =        NONE, PHASEE, PHASED,  
FFTEL, FFTER, FFTDL or FFTDR

### **MVN Print software version number**

This command is print the software version number of a thing.

See the ?? command.

**MVN ty** print software version number of thing ty

ty        =        ALL, CP, CPX,  
DSPD, DSPE, DSPV, DSPR  
DSPDX, DSPEX, DSPVX, DSPRX,  
DSPDXX, DSPEXX, DSPVXX, DSPRXX,  
TH0, TH1, TH2, TH3, TH4, TH5, TH6,  
TH7, TH8, TH9, TH10, TH11, TH12,  
TH13, TH14, TH15, TH16, TH17,  
TH18, TH19, TH20, TH21, TH22,  
TH23

### **MWP Set watch port**

453

This command is used to set the output RS232 port for debugging messages from internal processes. For example, each time a relay or cue message is sent or received, then a message is output to the

port issuing the command. The watch port, when enabled, allows a look at internal communication in the cdqPRIMA.

**MWP ?** print current watch port

**MWP wb** set watch port to wp to watch wb items

**wb** = ABCDEFGHIJKLMNOPQRSTUVWXYZ or  
NONE

A = decoder HF2=0 dsp interrupts  
B = encoder dsp interrupts  
C = reed soloman dsp interrupts  
D = vu dsp interrupts  
E = event to action results (action word)  
F = quiet detector scaled values from vu dsp  
G = quiet detector raw values from vu dsp  
I = decoder HF2=1 dsp interrupts  
J = messages to TA port  
K = messages from TA port  
L = decoder DSP host vector messages  
M = encoder DSP host vector messages  
N = reed soloman DSP host vector messages  
O = vu DSP host vector messages  
P = phase check in phase process  
Q = time code buffer to encoder  
R = time code buffer from decoder  
S = out going link word message  
T = incomming link word message  
U = peak detector scaled values from vu dsp  
V = peak detector raw values from vu dsp  
W = command from far end prima  
X = response to far end prima  
Y = command sent to far end prima  
Z = response from far end prima

### **HELP Print command summary for all commands**

This command is used to print a summary of all help commands.

See the CQQ, DQQ, EQQ and MQQ commands.

**HELP** print command summary

454

## WHAT IS CLAIMED IS:

1. A system for compressing and decompressing data bit streams, said system being comprised of:

at least one external input means for receiving audio data;

5 a plurality of external input means for receiving ancillary data bit streams;

a multiplexer capable of receiving said plurality of ancillary data bit streams, said multiplexer producing a composite bit stream of ancillary data;

10 an encoder, containing a compression technique, which receives said audio data and said composite ancillary data bit stream and produces a resulting compressed data bit stream;

at least one digital interface output module for externally outputting said compressed data bit stream;

15 at least one digital interface input module for externally inputting an external compressed data bit stream;

a decoder, containing a decompression technique for decompressing data produced by said encoder compression technique, wherein said decoder receives said external compressed data bit stream and produces decompressed audio and composite ancillary data  
20 bit streams;

a demultiplexer capable of receiving said decompressed composite ancillary data bit stream, said demultiplexer producing a plurality of separate decompressed ancillary bit streams;

at least one external output means for outputting said decompressed audio data;

a plurality of external output means for outputting said decompressed ancillary data bit streams.

5

2. An audio CODEC for providing high quality digital audio comprising:

an analog to digital converter for converting an analog audio signal to a digital audio bit stream;

10

an encoder for compressing said digital audio bit stream;

a decoder for decompressing said compressed digital audio bit stream;

an output allowing a user to monitor the digital audio output; and

15

at least one control for allowing said user to adjust said digital audio output.

3. A method for providing high quality digital audio comprising the steps of:

providing an input analog audio signal;

20

providing at least one psycho-acoustic parameters;

converting said input analog audio signal into a digital signal;

coding said digital signal in accordance with said at least one psycho-acoustic parameter;

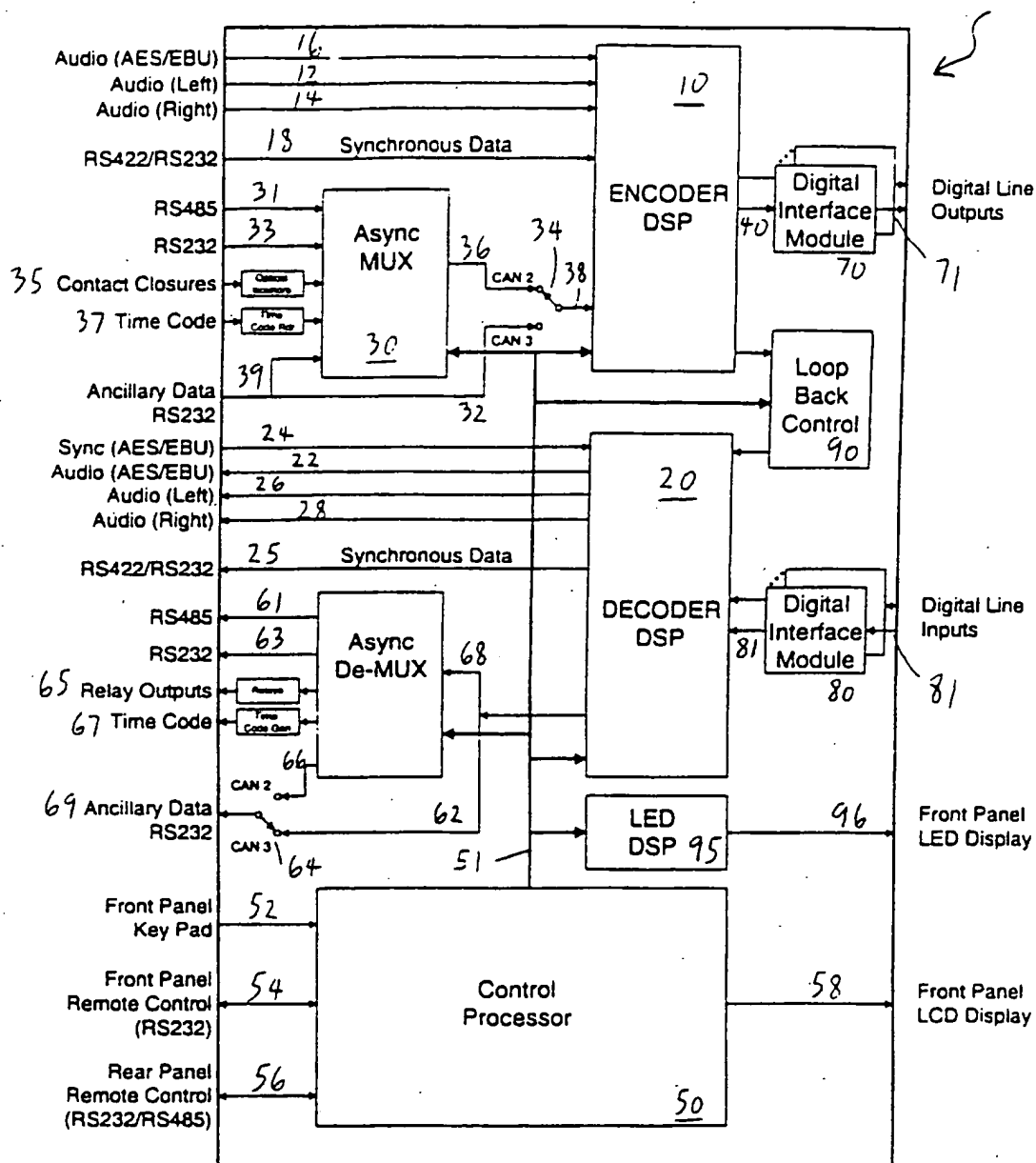


Figure 1

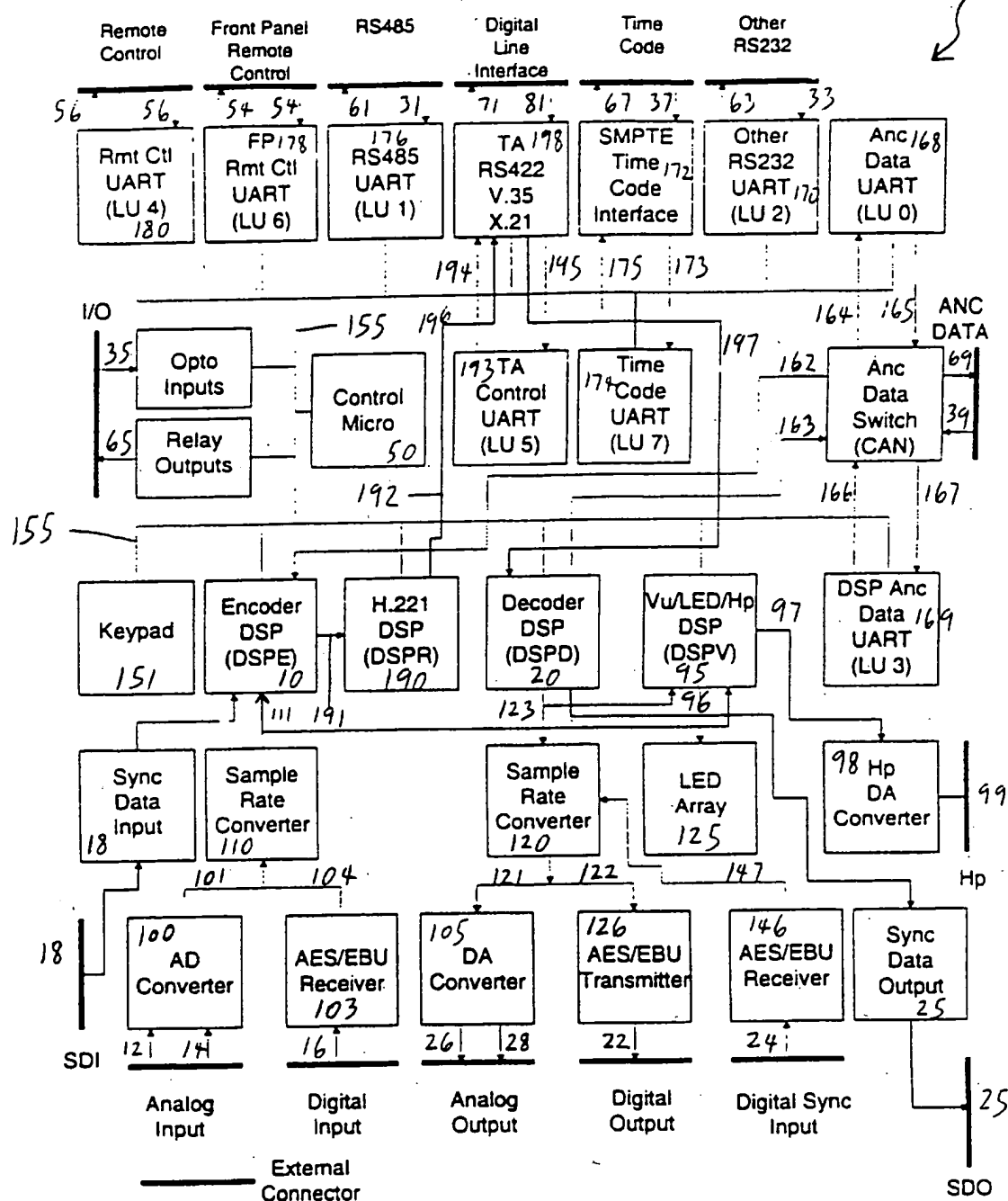


Figure 2



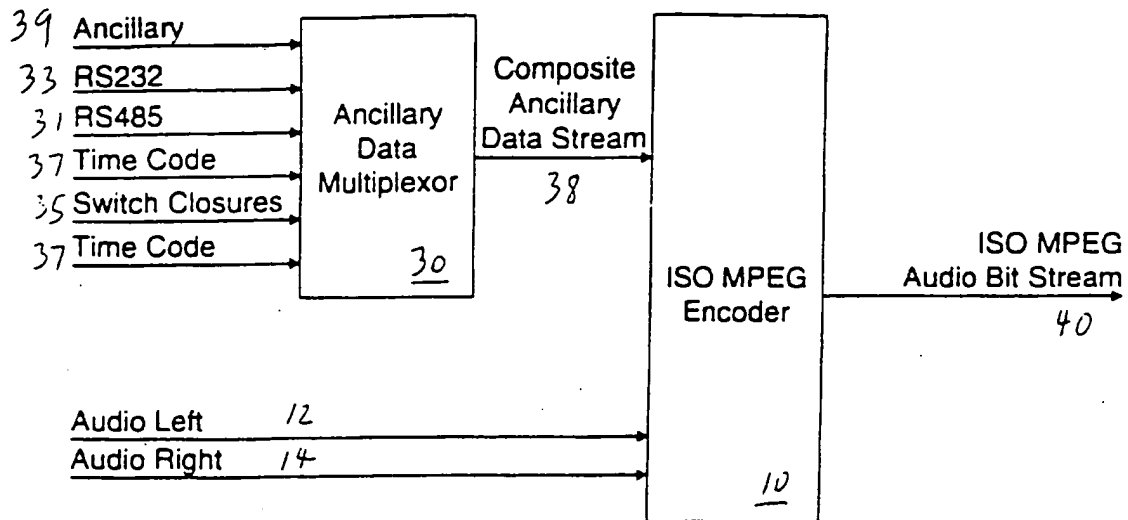


Figure 3

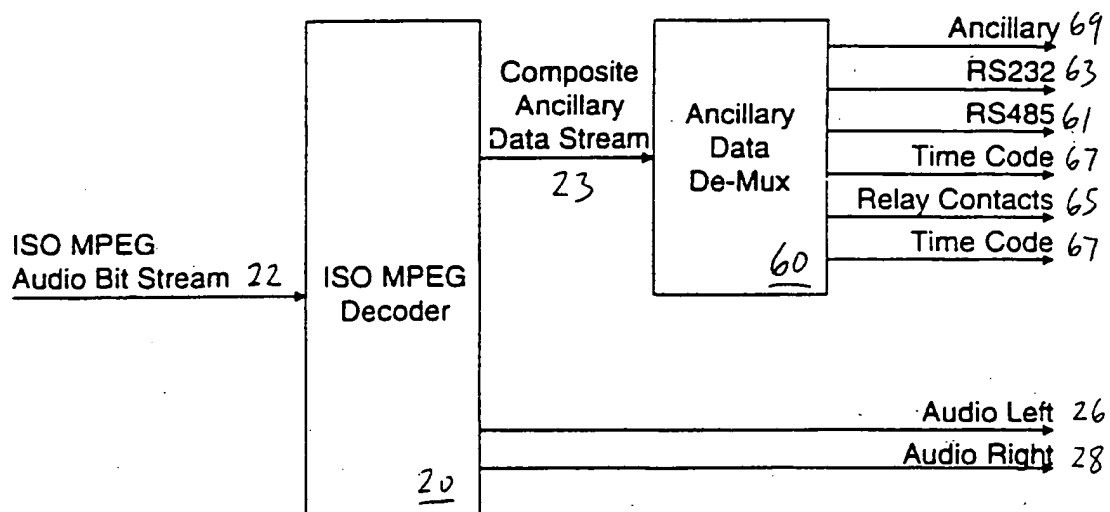


Figure 4

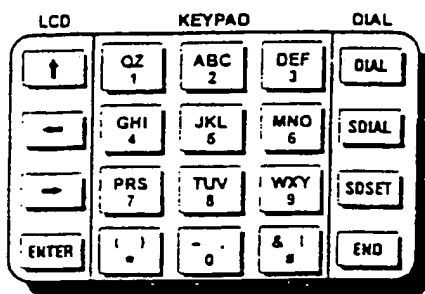


Figure 5

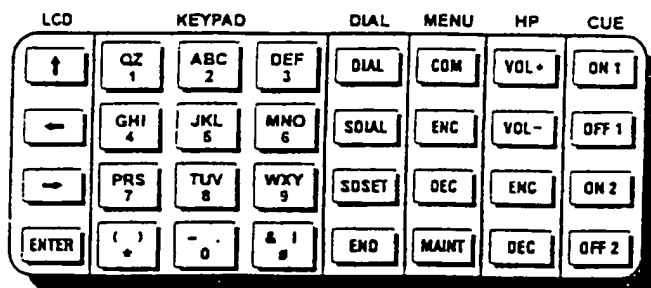


Figure 6

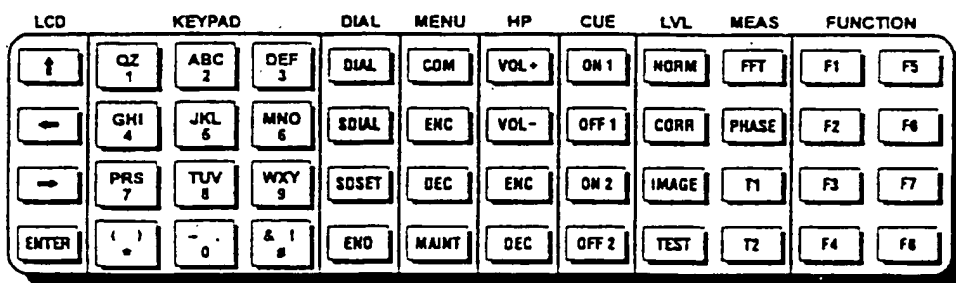


Figure 7

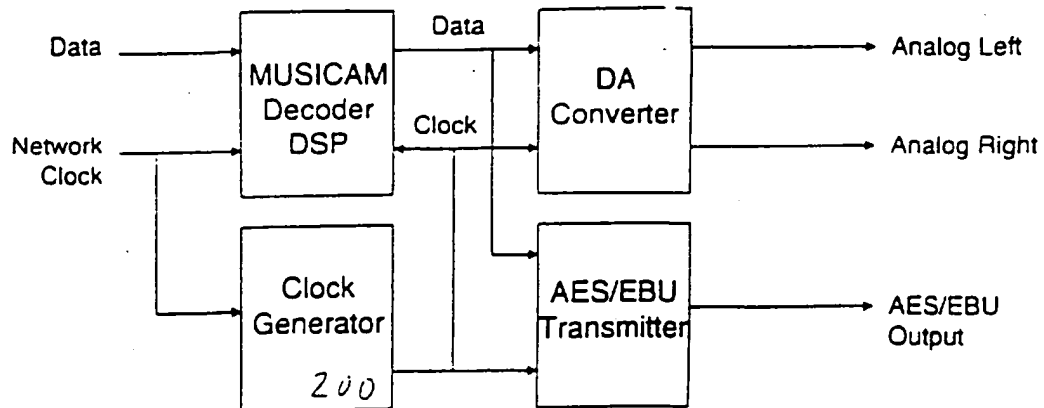


Figure 8

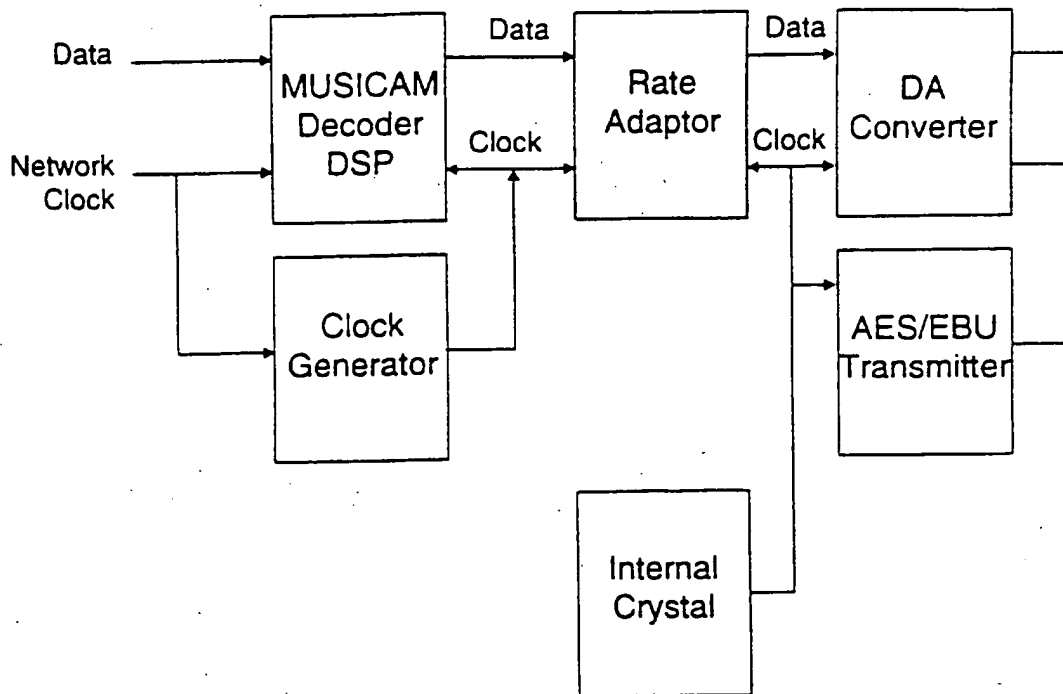


Figure 9

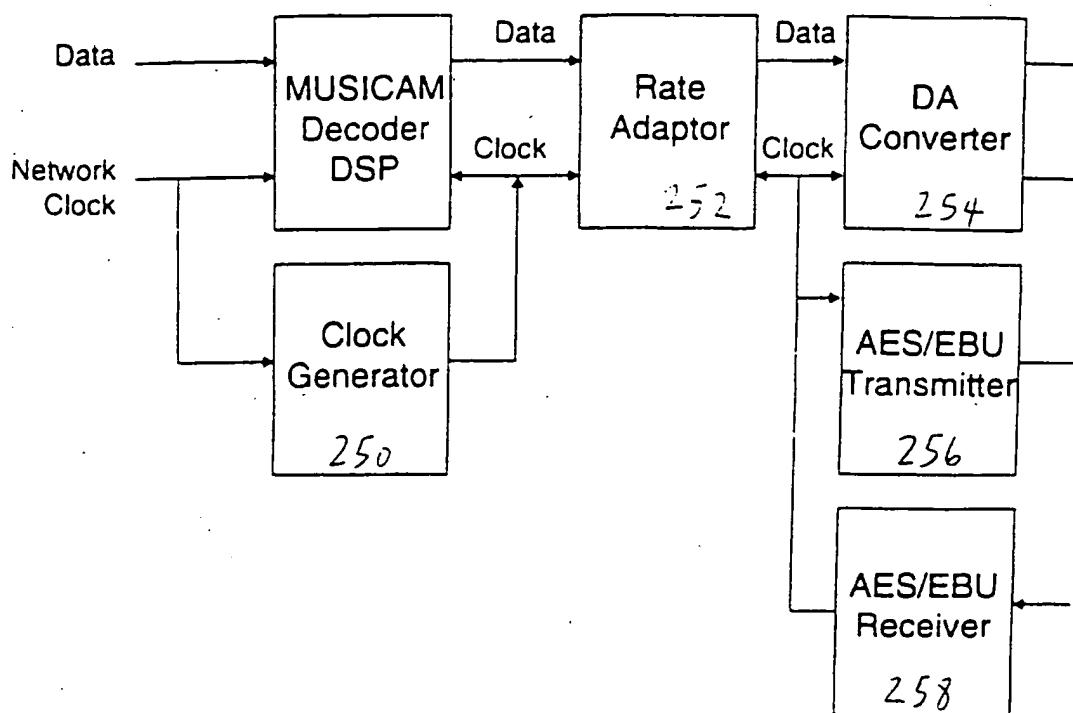


Figure 10

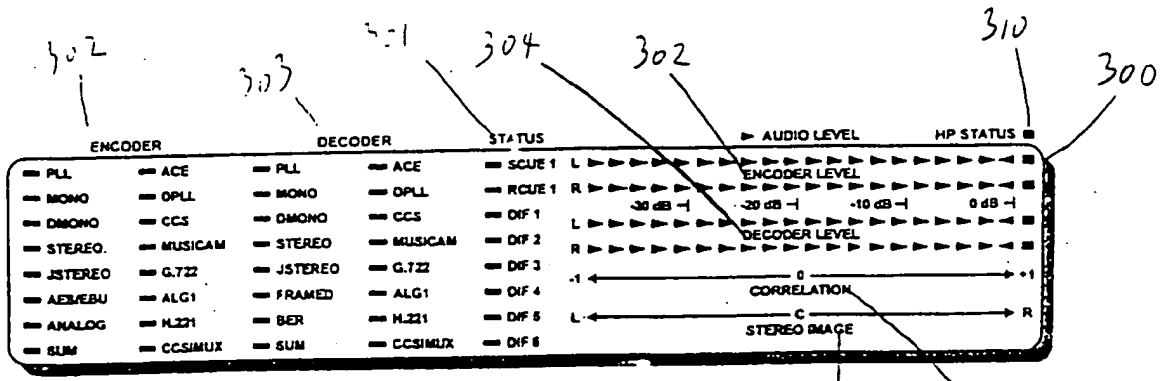


Figure 11

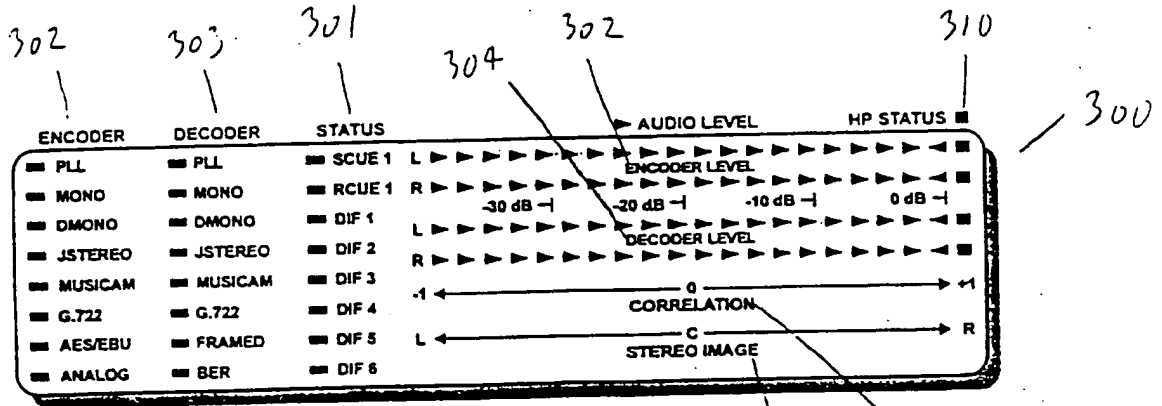


Figure 12

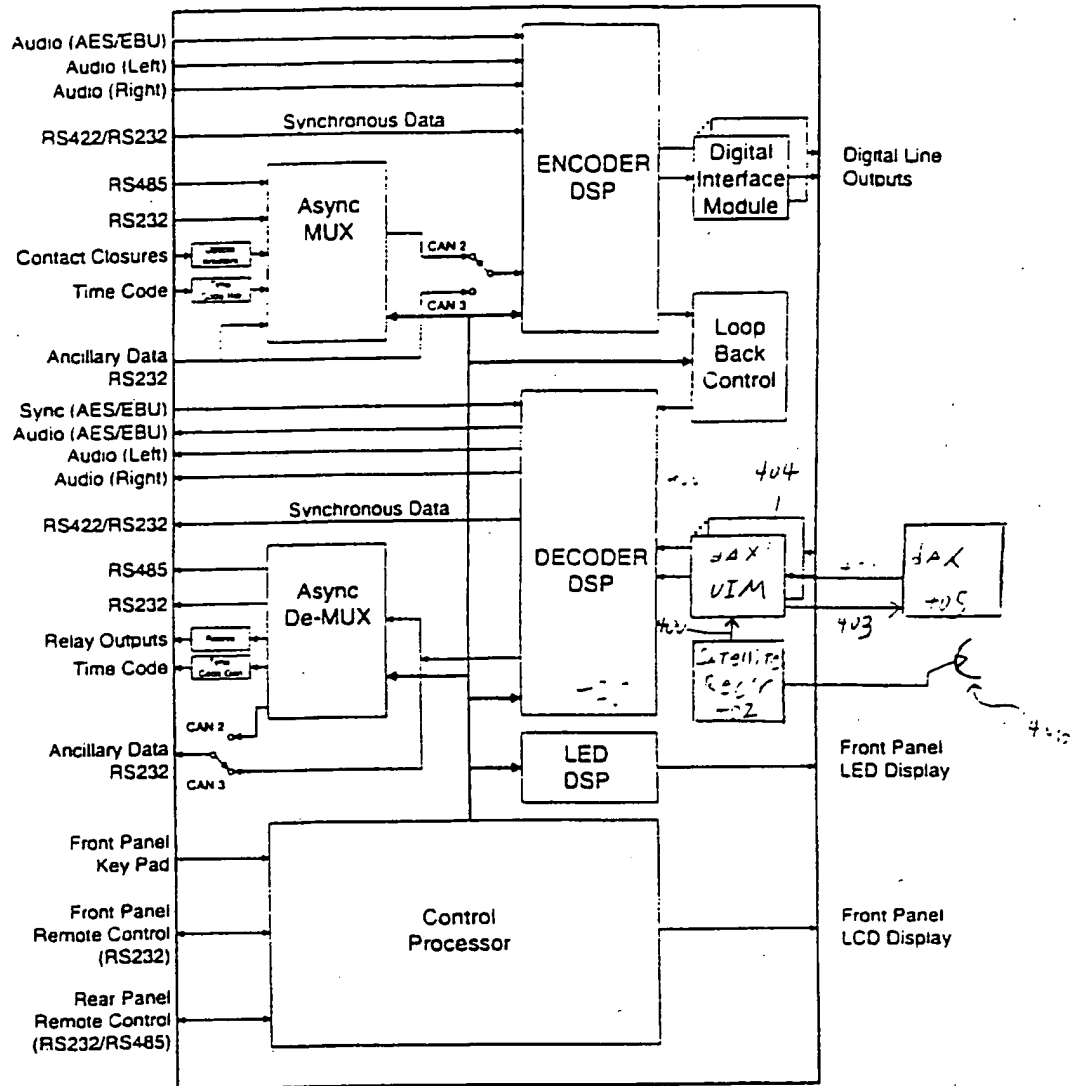


Figure 15

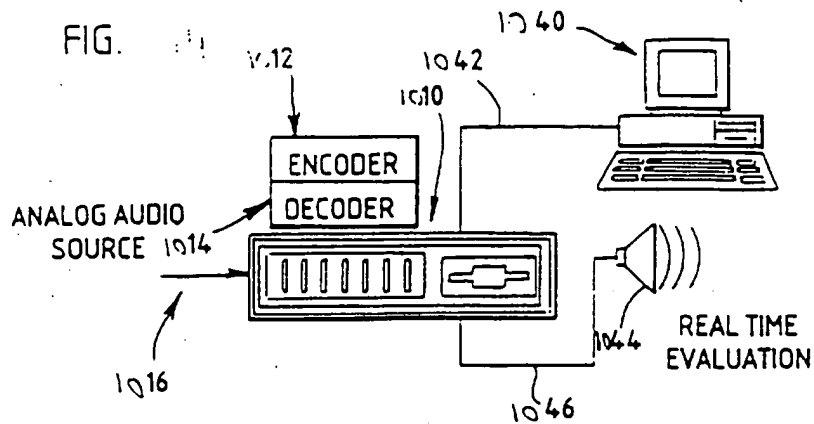
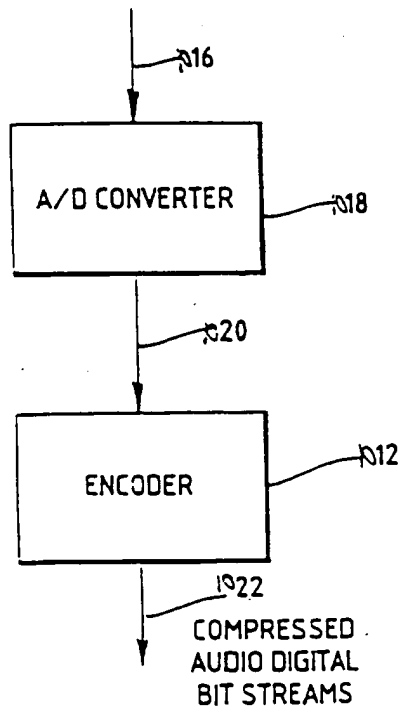
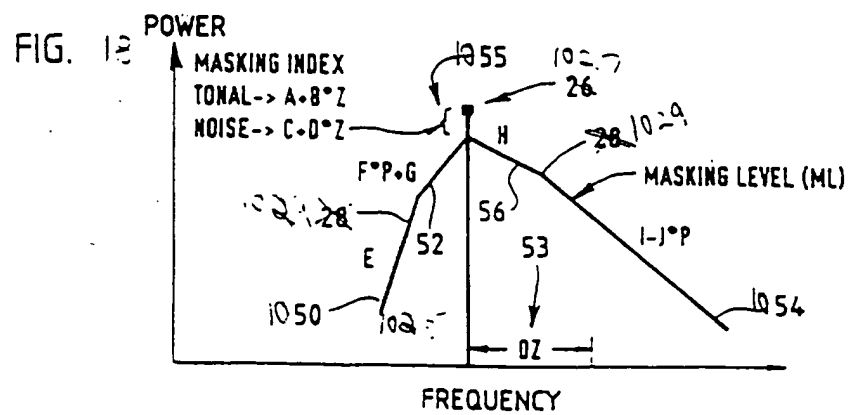
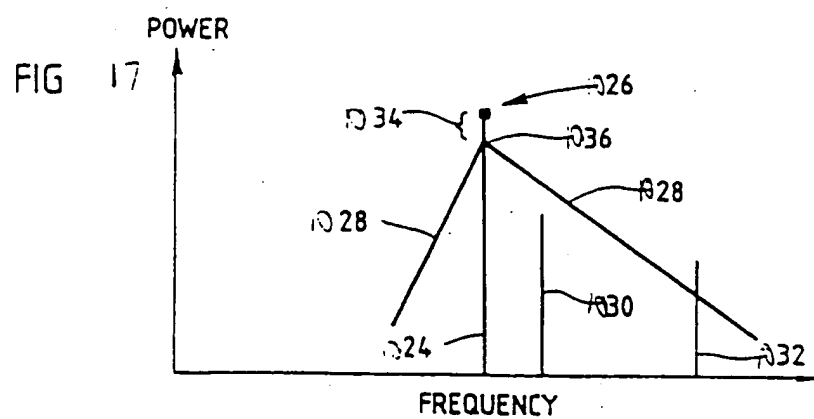
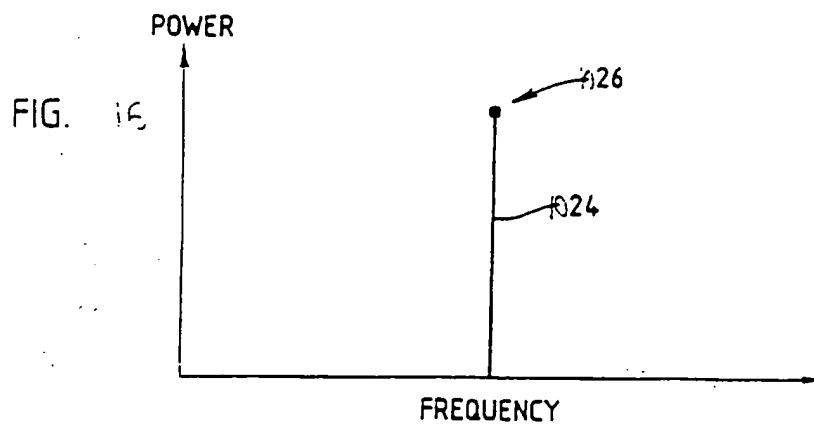


FIG. 15 ANALOG AUDIO INPUT SIGNAL







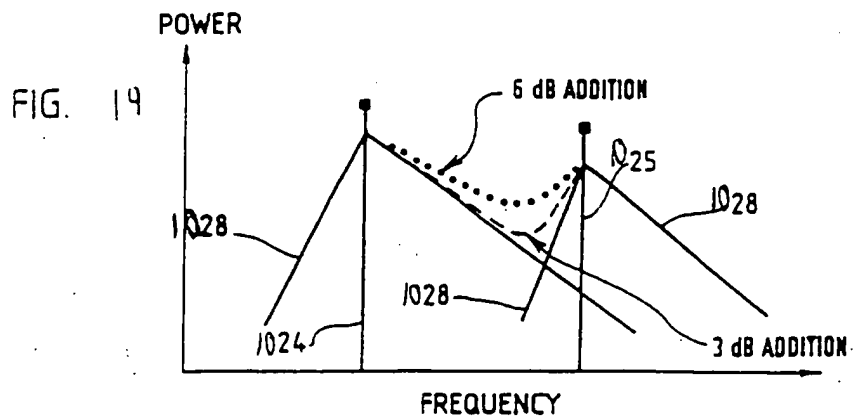
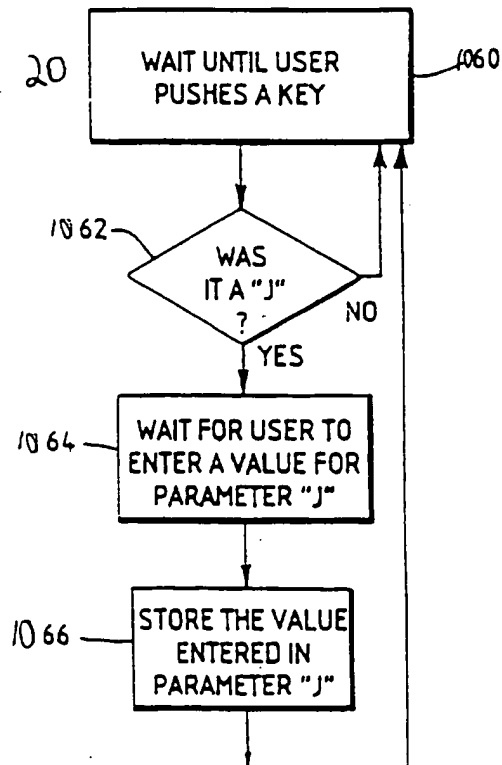
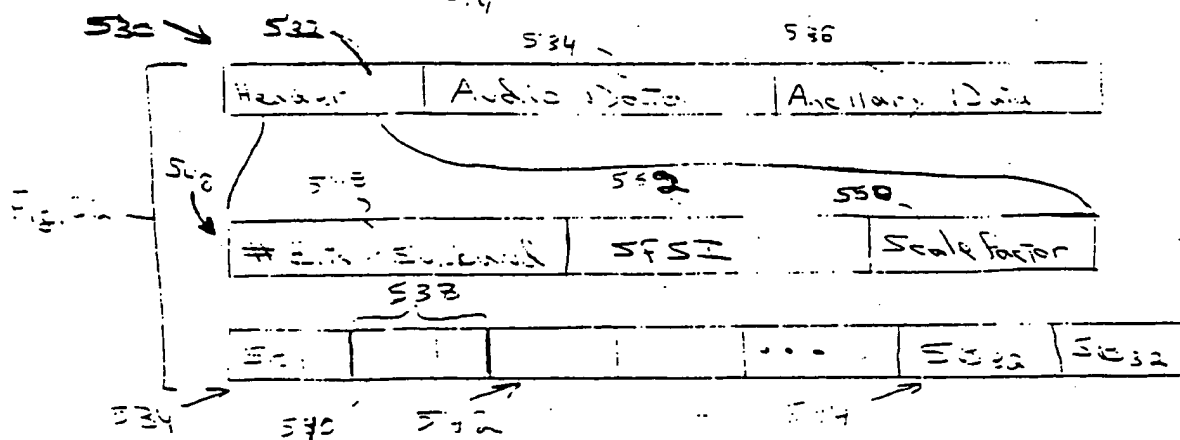
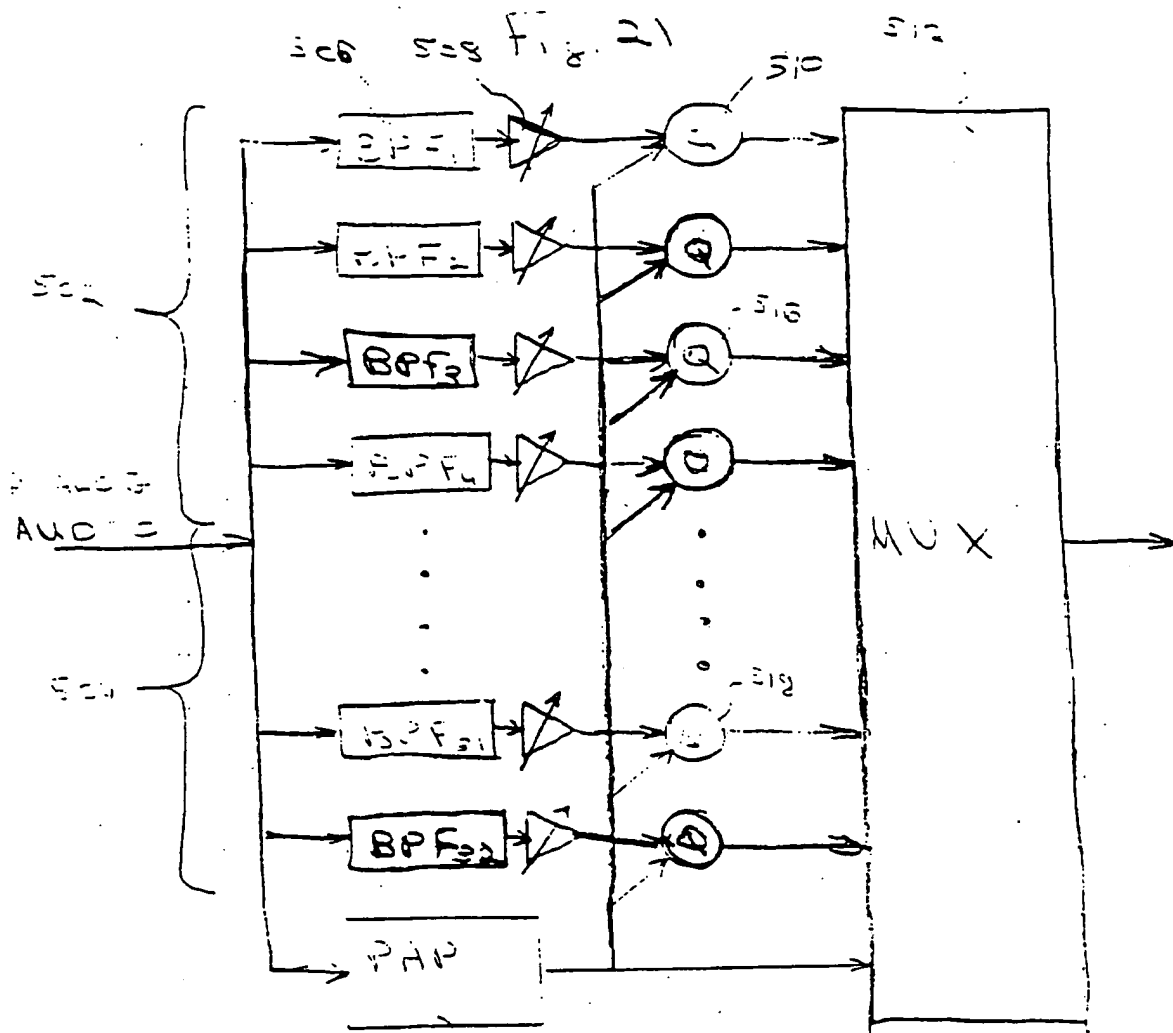
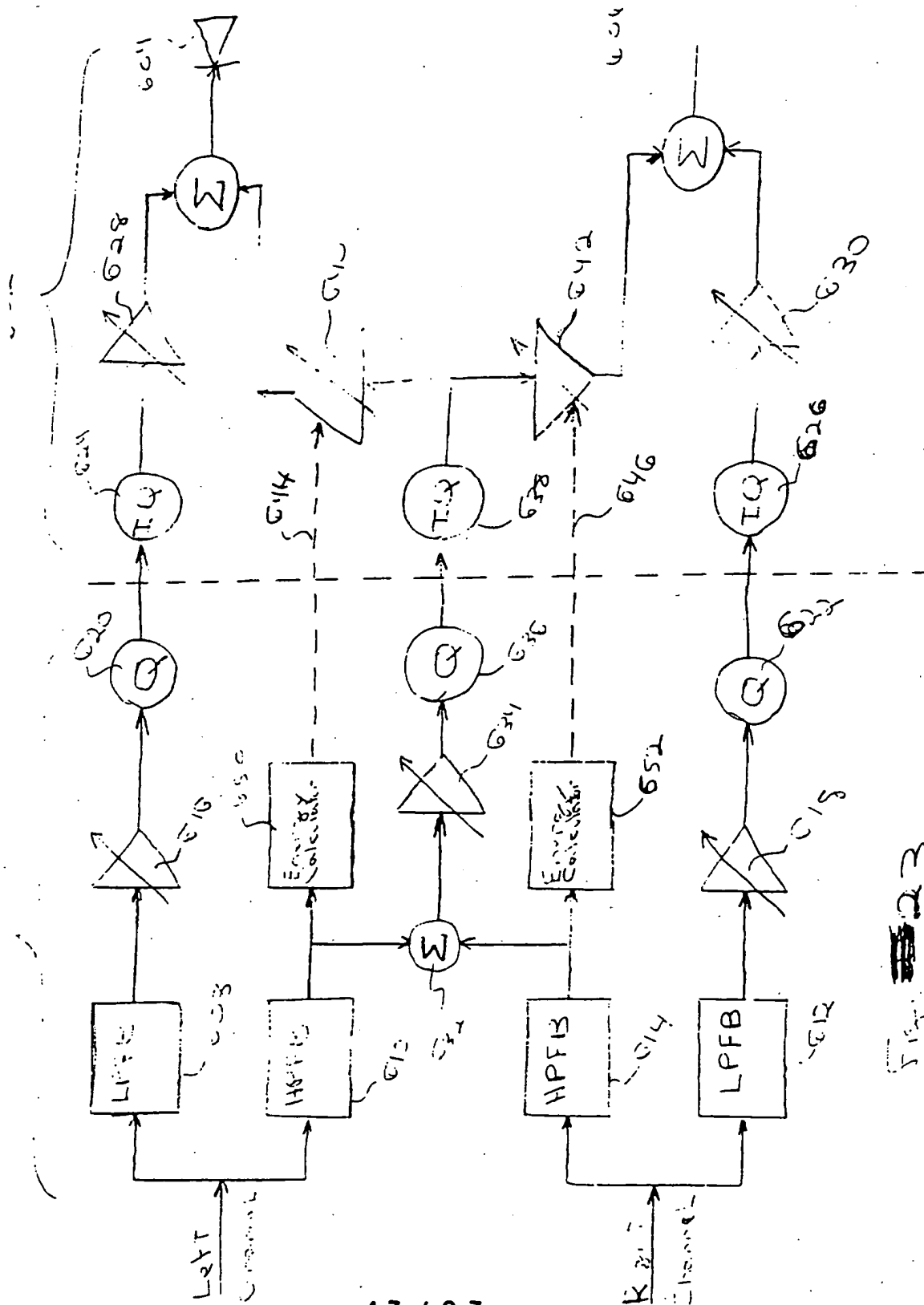


FIG. 20







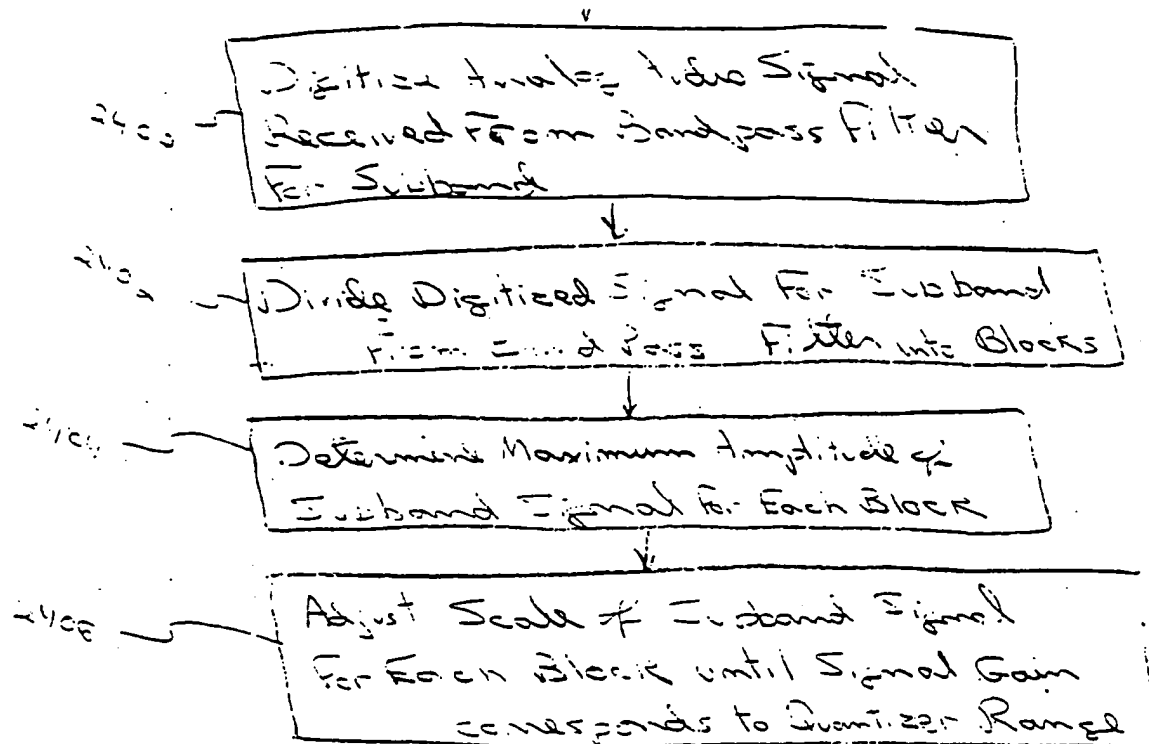


Fig. 24

Fig. 25A

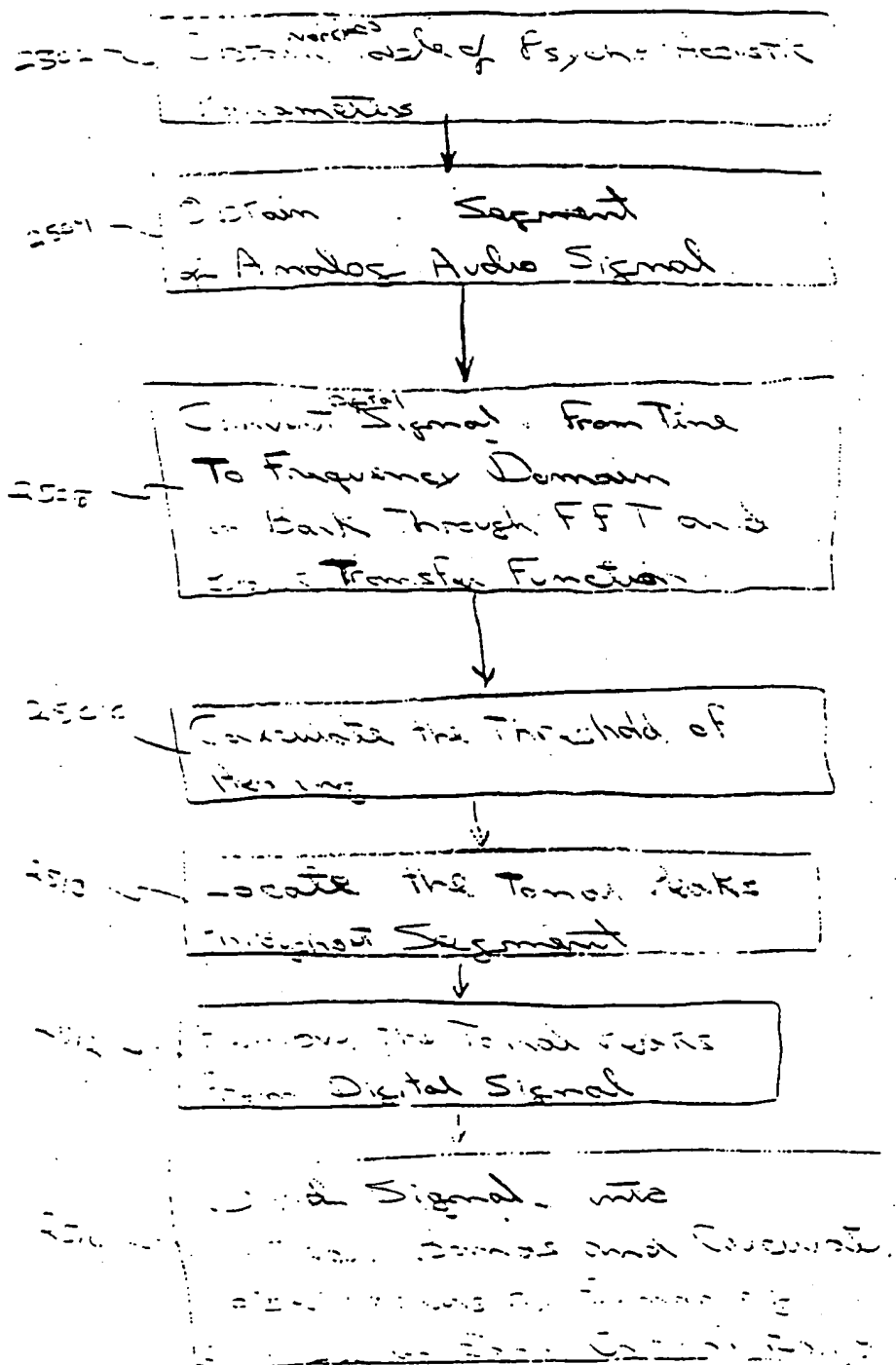


Fig. 25B

From Fig. 25A

2518

Calculate Masking Skirts for  
 Total and Noise Increase based  
 on noise levels A-T and Application  
 and Thresholds of Maskers

2520

Noise optional

Calculate Masking Skirts and  
~~Calculate Masking Skirts and~~

the increase of hearing to obtain

2522

the Global Masking Threshold

Calculate GHT noise levels to calculate

the hearing of hearing and least hearing

the hearing of GHT noise levels to calculate

2524

Assign noise level level for

the hearing of hearing and least hearing

the hearing of hearing and least hearing

the hearing of hearing and least hearing



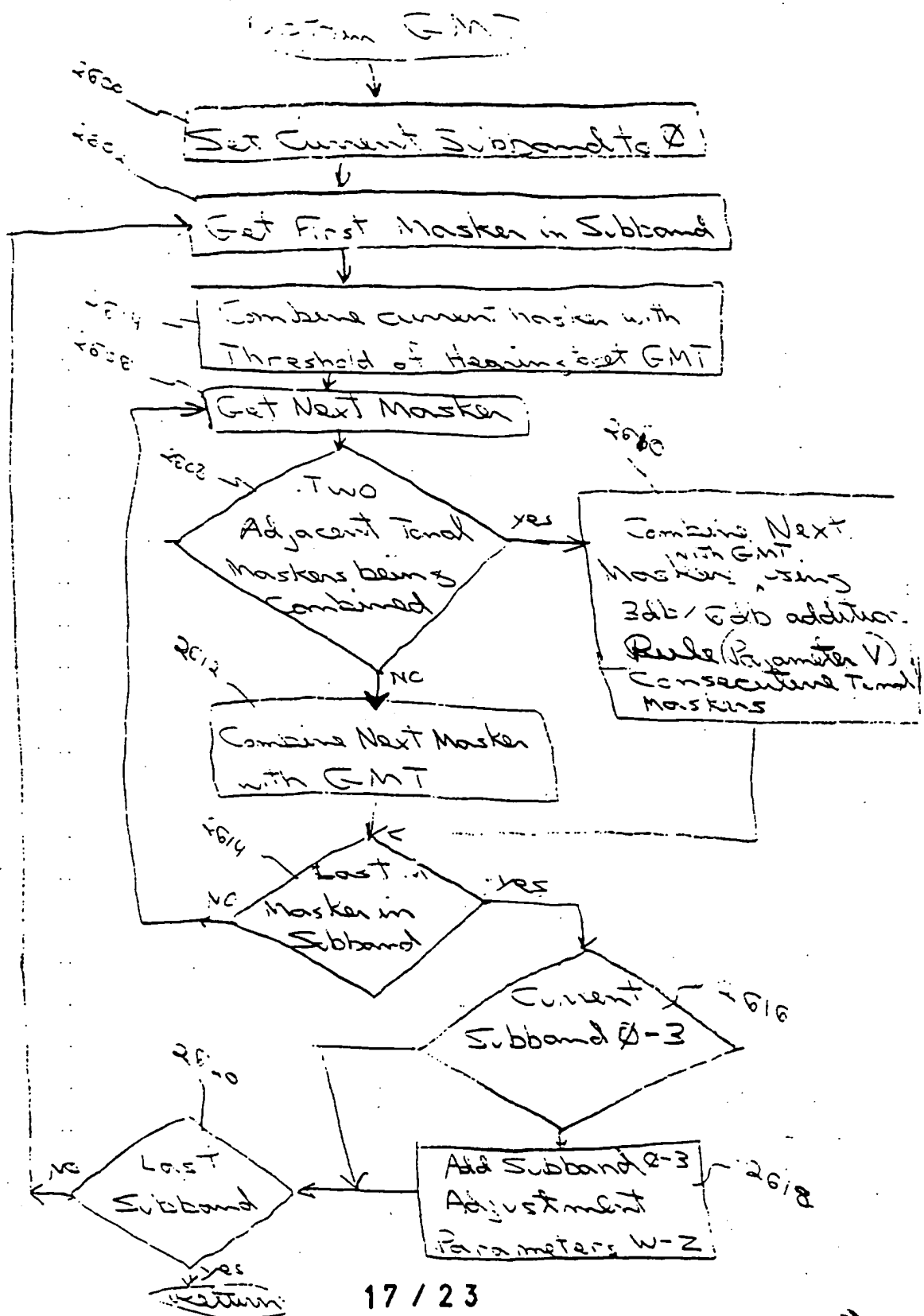
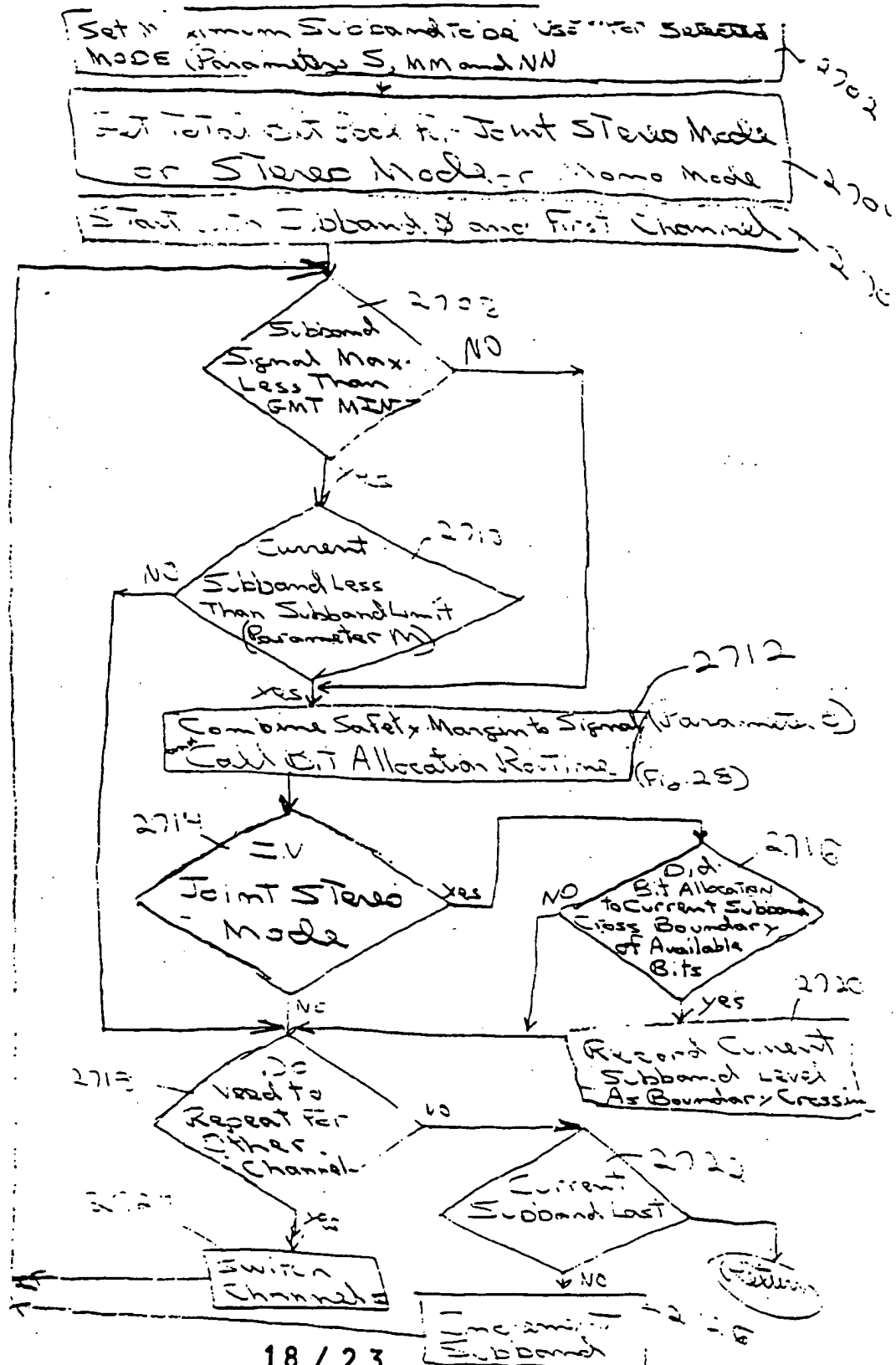
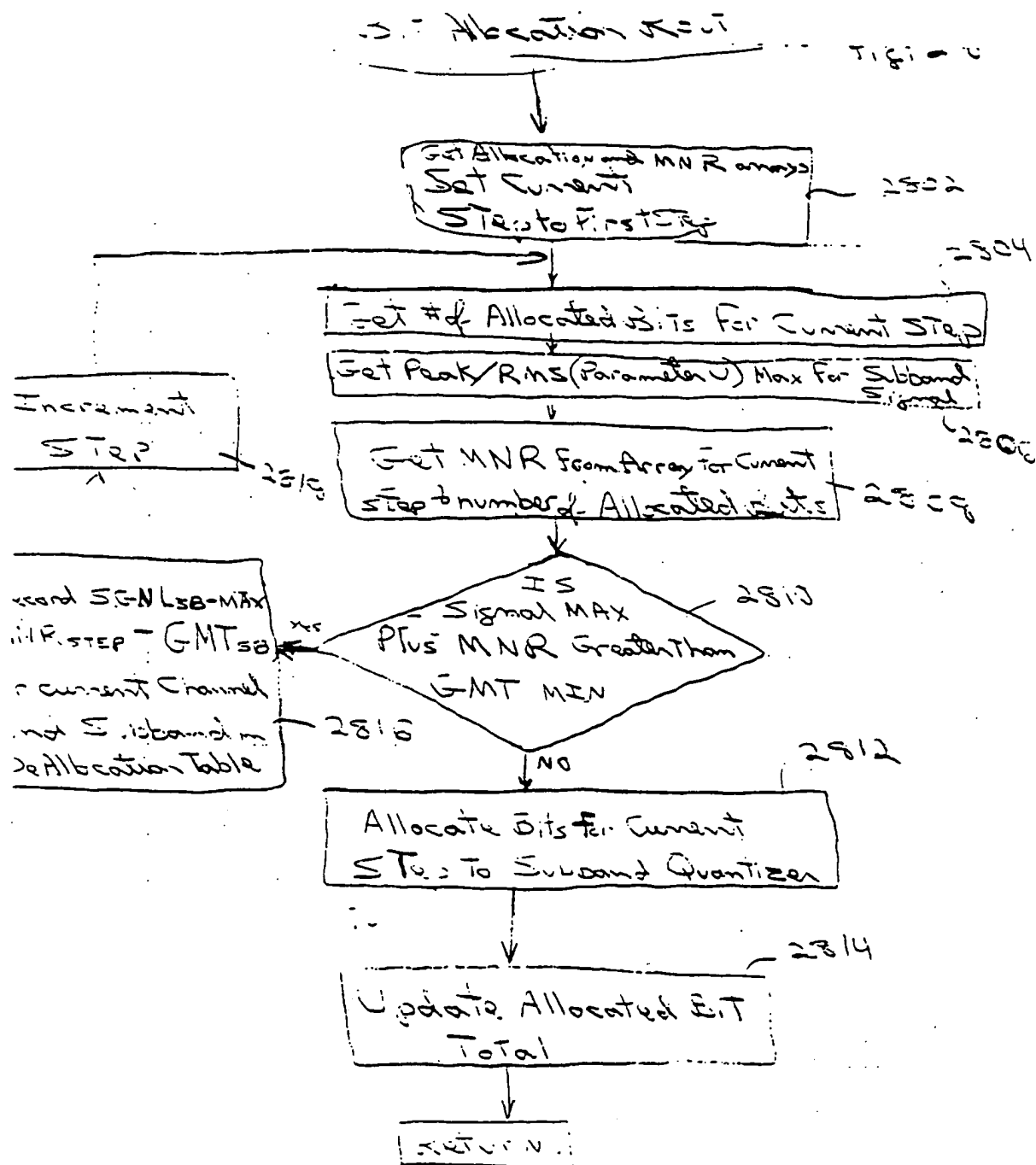


FIG. 27







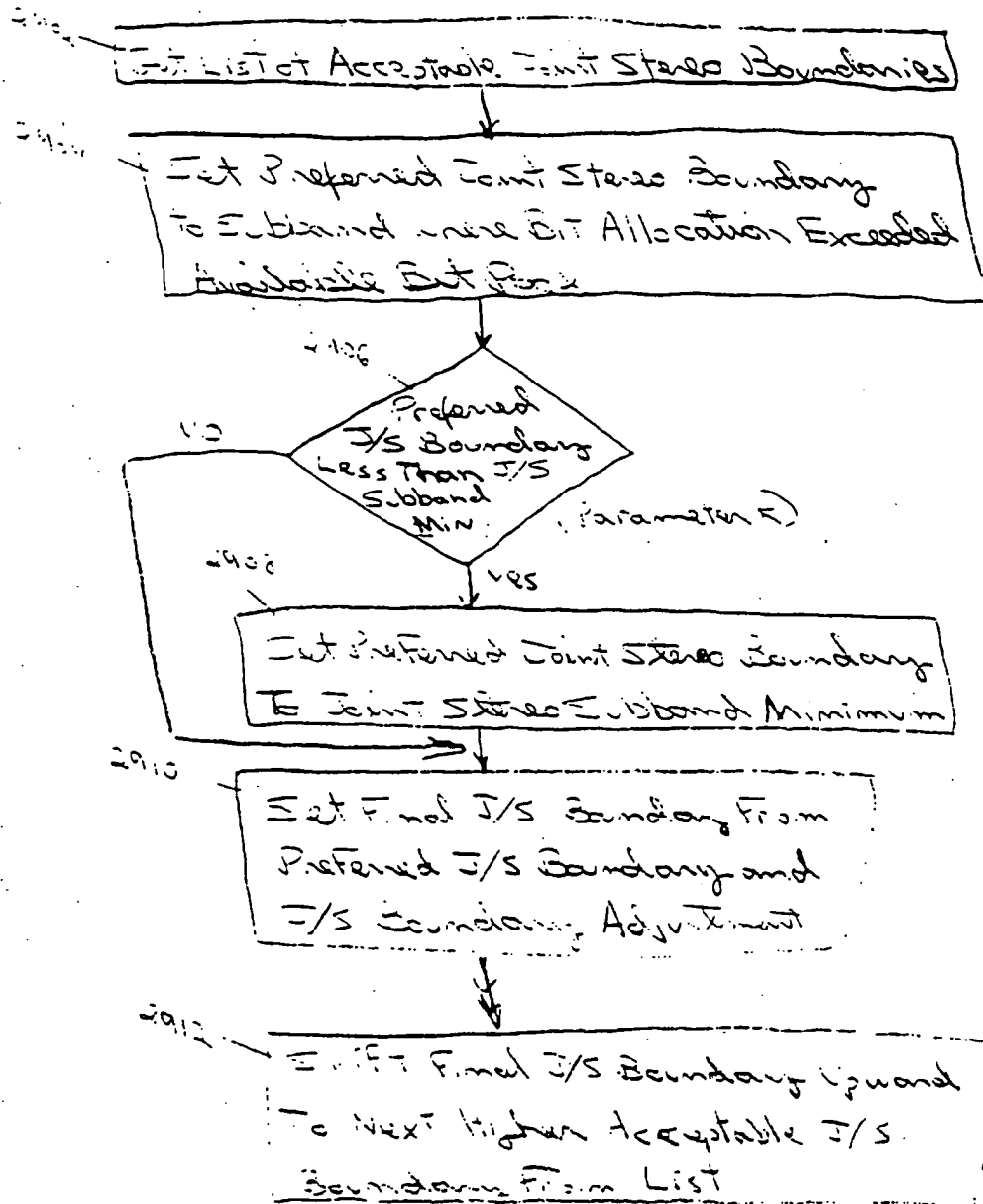
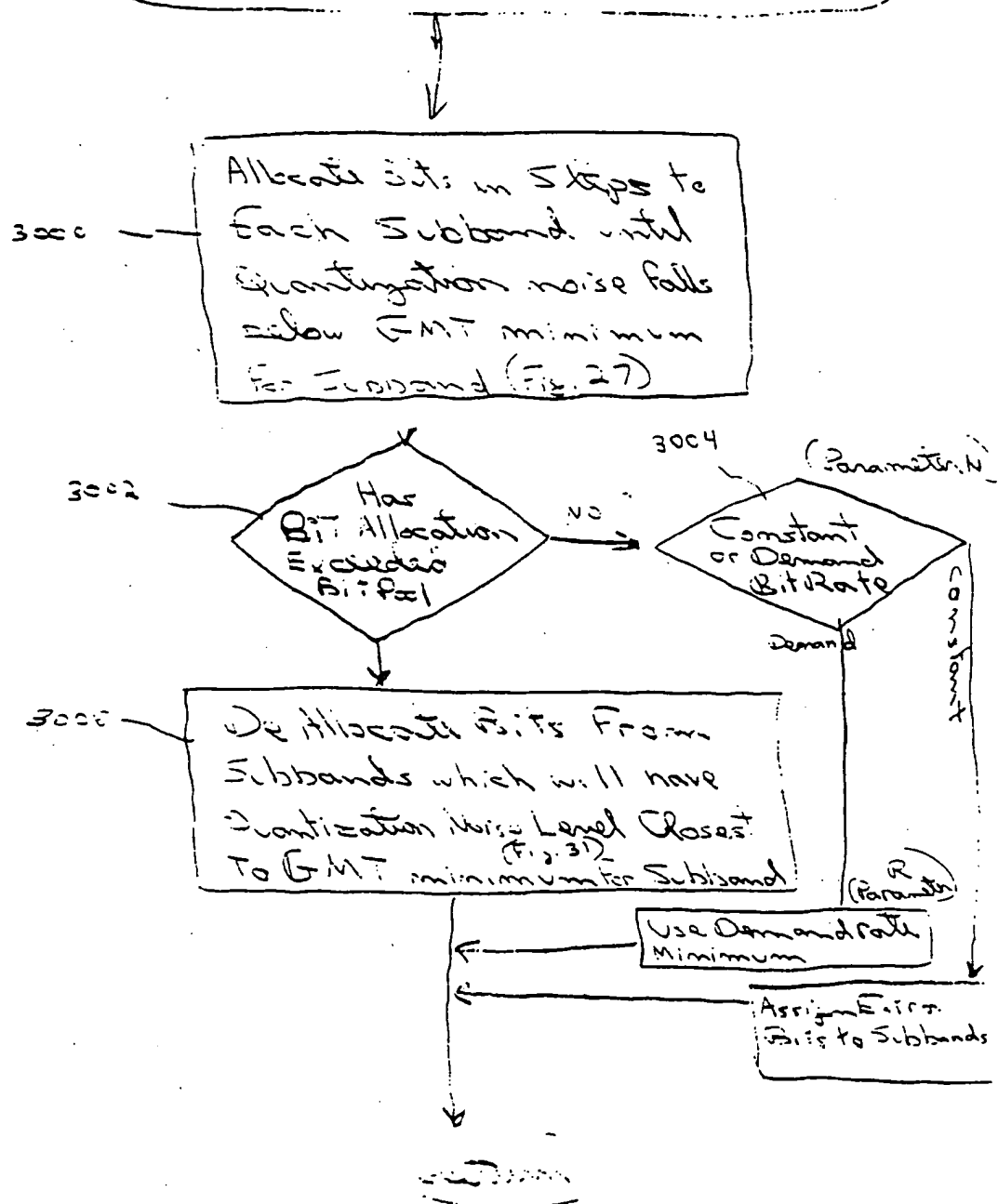
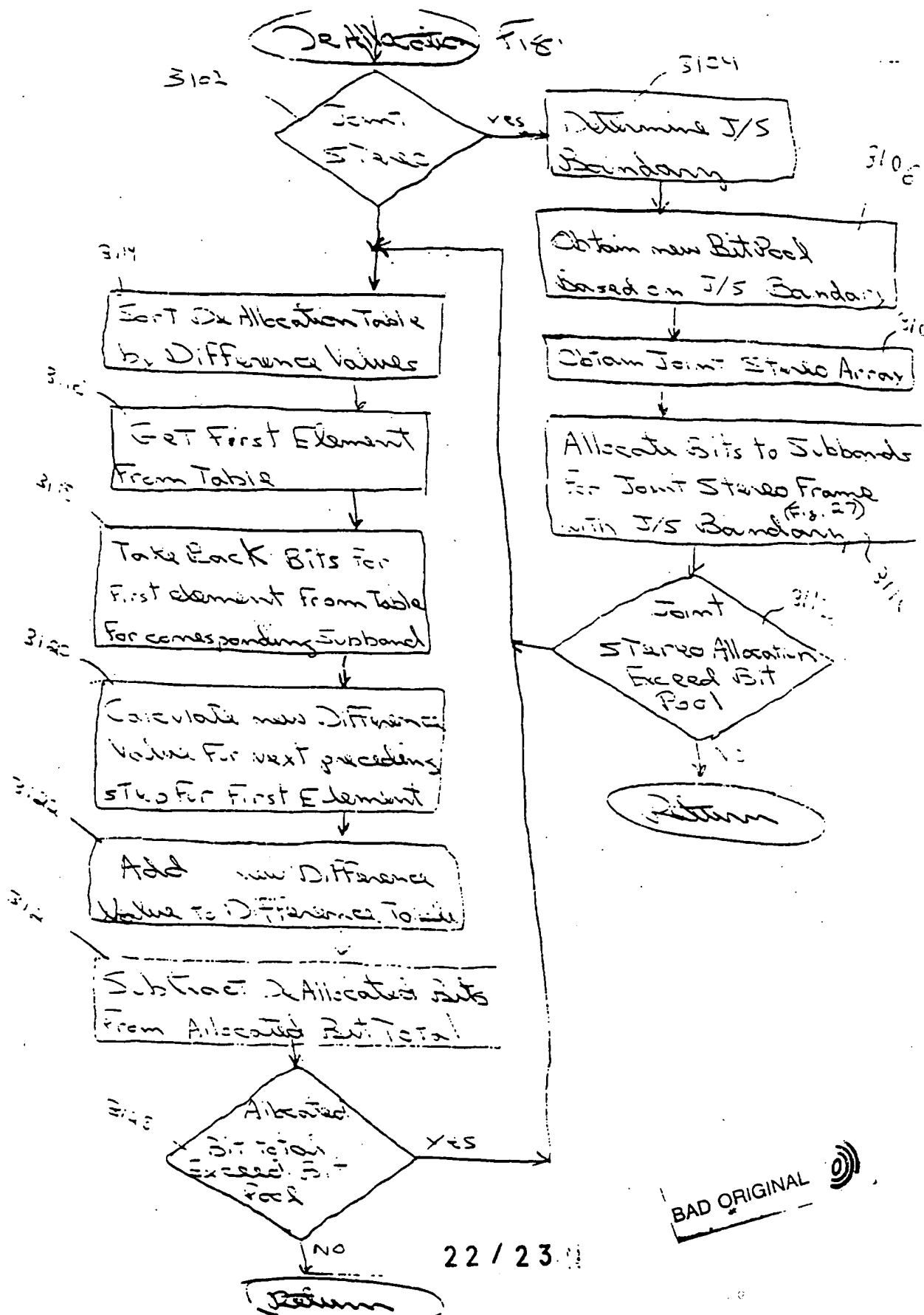


Fig. 24



2522 (A551610) Quantization level





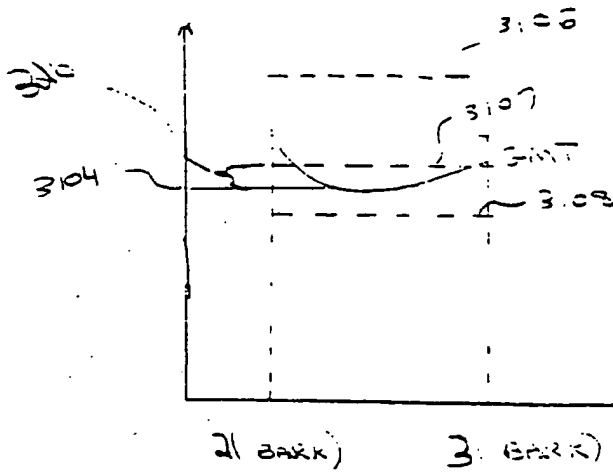


Fig. 32a

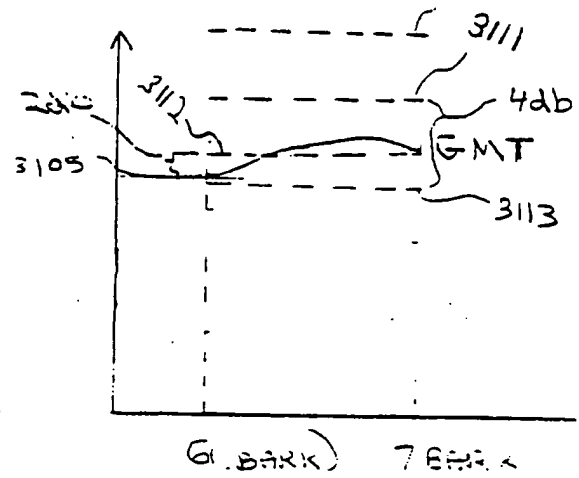


Fig. 32b

Fig. 33

Channel	Subband	Difference Value	Allocation STEP
2	7	2 db	3
2	3	3 db	2
	7	4 db	2

# INTERNATIONAL SEARCH REPORT

International application No.

PCT/US96/04974

## A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :G10L 3/00

US CL :395/2.1

According to International Patent Classification (IPC) or to both national classification and IPC

## B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 395/2.1, 2.12, 2.14, 2.2, 2.34, 2.38, 2.39

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS: speech or audio, ancillary, auxillary, psycho-acoustic

## C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y,E	US, A, 5,530,655 (LOKHOFF ET AL) 25 June 1996, Abstract, Figures 12 and 16.	1-3
X,E	US, A, 5,515,107 (CHIANG ET AL) 07 May 1996, abstract, figures 1-3, 4E.	1, 3
Y,E		2
Y	US, A, 5,161,210 (DRUYVESTYEN ET AL) 03 November 1992, abstract, figure 1.	1-3
Y,P	US, A, 5,493,647 (MIYASAKA ET AL) 20 February 1996, abstract.	1-3
Y	US, A, Re32,124, (ATAL) 22 April 1986, abstract, figures 1, 2.	2

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:	*T* later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
*A* document defining the general state of the art which is not considered to be part of particular relevance	*X* document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
*E* earlier document published on or after the international filing date	*Y* document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
*L* document which may throw doubt on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	*G* document referring to an oral disclosure, use, exhibition or other means
*O* document referring to an oral disclosure, use, exhibition or other means	*P* document published prior to the international filing date but later than the priority date claimed
*P* document published prior to the international filing date but later than the priority date claimed	*G* document member of the same patent family

Date of the actual completion of the international search

14 AUGUST 1996

Date of mailing of the international search report

10 SEP1996

Name and mailing address of the ISA/US  
Commissioner of Patents and Trademarks  
Box PCT  
Washington, D.C. 20231

Facsimile No. (703) 305-3230

Authorized officer

DAVID D. KNEPPER

Telephone No. (703) 305-9600

**This Page is Inserted by IFW Indexing and Scanning  
Operations and is not part of the Official Record**

**BEST AVAILABLE IMAGES**

Defective images within this document are accurate representations of the original documents submitted by the applicant.

Defects in the images include but are not limited to the items checked:

- ☐ BLACK BORDERS
- ☐ IMAGE CUT OFF AT TOP, BOTTOM OR SIDES
- ☒ FADED TEXT OR DRAWING
- ☐ BLURRED OR ILLEGIBLE TEXT OR DRAWING
- ☐ SKEWED/SLANTED IMAGES
- ☐ COLOR OR BLACK AND WHITE PHOTOGRAPHS
- ☐ GRAY SCALE DOCUMENTS
- ☐ LINES OR MARKS ON ORIGINAL DOCUMENT
- ☐ REFERENCE(S) OR EXHIBIT(S) SUBMITTED ARE POOR QUALITY
- ☐ OTHER: \_\_\_\_\_

**IMAGES ARE BEST AVAILABLE COPY.**

As rescanning these documents will not correct the image problems checked, please do not report these problems to the IFW Image Problem Mailbox.